Folks,

Naval Sonar, NAVPERS 10884, 1953 was created a few years after WW II and incorporates the major invovations of WW II. It describes the peak of WW II US sonar technology with a hint of the comming Cold War innovations.

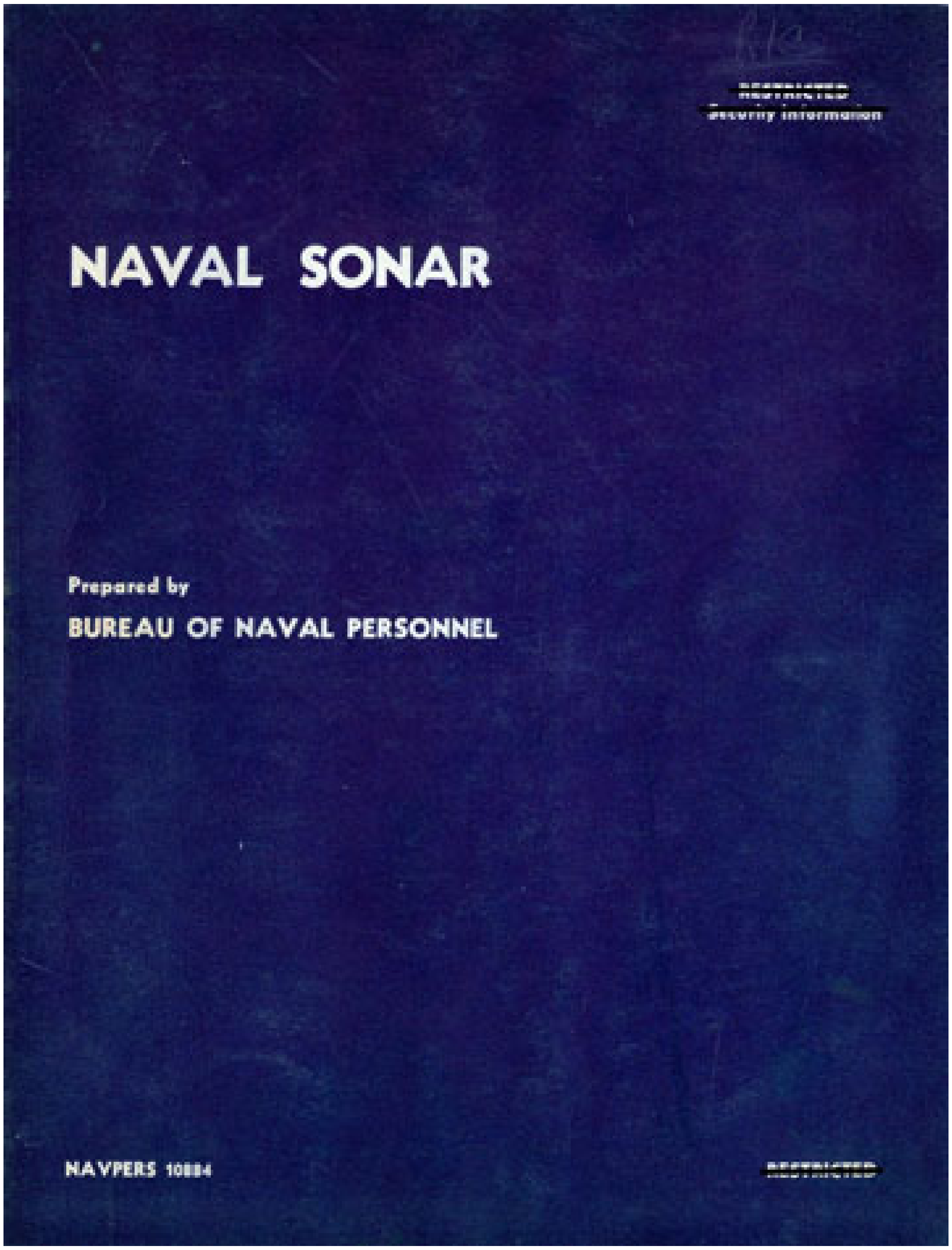
This manual has a bit more math and theory than many of the operator manuals we have presented on this web site. If you are not familiar with the math, please do not be discouraged, instead try skipping what you do not understand and move on. The farther into the manual you go the less theory and the more practical systems are described. At the end, there are chapters that describe the most successful systems with very little math.

In this online version of the manual we have attempted to keep the flavor of the original layout while taking advantage of the Web's universal accessibility. Different browsers and fonts will cause the text to move, but the text will remain roughly where it is in the original manual. In addition to errors we have attempted to preserve from the original this text was captured by optical character recognition. This process creates errors that are compounded while encoding for the Web.

Please report any typos, or particularly annoying layout issues to info@hnsa.org for correction.

Richard Pekelney
Webmaster

Search Naval Sonar



NAVAL SONAR

Prepared by
BUREAU OF NAVAL PERSONNEL

1953

NAVPERS 10884

PREFACE

Naval Sonar was prepared primarily as a text for a correspondence course for use in training Naval Reserve electronics officers. The aim in preparing this book has been to present a general over-all view of underwater sound principles and sonar equipment. Naval Sonar will serve as background information for officer personnel and advanced enlisted ratings who may be called upon to learn the details of installation, operation, and maintenance of sonar equipment.

The summary technical report of Division 6, NDRC, volume 7, Principles of Underwater Sound, and various sonar manufacturers' instruction books were used as primary reference sources for Naval Sonar.

This publication has been prepared by the U. S. Navy Training Publications Center for the Bureau of Naval Personnel.

CONTENTS

Foldout Illustrations:

Figure 13-3 -Circuit of the JP-1 audio amplifier with line filter.

Next Chapter

Search Naval Sonar

Match: All Format: Long Sort by: Score

Search: [ ] Search

Return to the Historic Naval Ships Association home page.

Version 1.00, 23 Oct 05

CHAPTER 1

BASIC PRINCIPLES OF UNDERWATER SOUND

Introduction

One of the most critical problems encountered by the Allies early in World War II was the submarine menace. Almost five thousand merchant ships were sunk and more than twenty million tons of war supplies were lost by enemy action. The struggle against enemy submarines was successful because we were able to detect and locate them whether they were surfaced, submerged, underway, or lying in wait.

The majority of enemy submarines attacked were detected and located by *sonar*. To date, sonar has been the most effective method of detecting completely submerged submarines. Other methods such as radio, radar, and infrared, have proved ineffective because their range of transmission in sea water is practically nil.

The development of SONAR during the period between the wars was an unspectacular, slow but steady conquest over the physical elements of the sea, culminating during and since World War II in one of the Navy's largest research and development programs. The word "SONAR" abbreviates *SO*und, *N*avigation, And *R*anging, and includes all types of underwater sound devices used for listening, depth indication, echo ranging, ship-to-ship underwater communication, and other uses. The importance of sonar in naval warfare cannot be overemphasized.

This text is divided into two general parts-(1) a brief discussion of the physics of sound propagation in an ideal medium, followed by a presentation of the peculiarities and limitations of sea water as a medium for the transmission of sound, and (2) a general study of the design and function of representative sonar equipments.

In planning this text it has beep assumed that the reader has a knowledge of elementary physics, mathematics, and electronics.

Characteristics of Sound in an Ideal Medium

IDEAL MEDIUM

The peculiarities and limitations of sound transmission in sea water are understood more easily if sound is first thought of as being transmitted in an ideal medium. Such a medium is assumed to be homogeneous, infinite in extent, and perfectly elastic. A homogeneous medium has the same properties throughout, such as temperature, pressure, salinity, and density. The infinite extent of the medium permits omission of boundary reflections. Perfect elasticity means that

SOUND AND PRESSURE WAVES

Most wave motion can be classified as either longitudinal or transverse. Sound waves are longitudinal and are characterized by the vibrating particles of the medium moving forward and backward parallel to the direction in which the waves are propagated.

The waves are composed of alternate compressions and rarefactions in the medium. The term "sound" is used in two senses-subjectively, it denotes the

the medium, when distorted or displaced, returns to equilibrium with no loss of energy by internal friction. It is obvious that the properties of sea water differ greatly from those of an ideal medium. Nevertheless, a discussion of sound transmission in the two media is advantageous.

auditory sensation experienced by the ear, and objectively, it denotes the vibratory motion which gives rise to that sensation. This motion is often called the *stimulus*. All stimuli do not produce sensations of hearing, because the average ear responds to sounds in the frequency range from approximately 16 cycles per second to

1

15,000 cycles per second. This range is known as the sonic frequency range. Frequencies above 15,000 cycles per second, although not within the range of response of the average ear, are useful and can be detected with proper instruments. These frequencies are known as *ultrasonic* frequencies. The dividing line is not sharply defined; many people-particularly young persons-can hear above 15,000 cycles per second; but some standard must be adopted, and 15,000 cycles per second is used as the arbitrary dividing line. Before rockets and aircraft attained speeds greater than the speed of sound, frequencies above 15,000 cycles per second were designated as supersonic. However, it is now agreed that the term "supersonic" designates velocities greater than the velocity of sound and the term "ultrasonic" means frequencies above 15,000 cycles per second.

EQUATION OF WAVE MOTION

The *period*, T, of a vibrating particle in a medium is the time in which it completes one vibration, and the *frequency*, F, is the number of vibrations completed per second. Frequency is expressed as "cycles per second (cps)," "kilocycles (kc)," and "megacycles (mc)." In such units "cycles" is understood to mean "vibrations per second." The maximum displacement from the undisturbed position is called the amplitude of vibration.

Two wave motions vibrating with the same frequency have definite *phase* relations. They are to its wavelength, λ, so the velocity of the wave is $v=\lambda/T$. Because the period, T, is the reciprocal of the frequency, F, it follows that the wave velocity is

$$v=F\lambda. \quad (1\text{-}1)$$

In this equation the wave velocity, v, is determined completely by the properties of the transmitting medium and is independent of the frequency of the source and of the wavelength. When F changes there must be a corresponding change in λ so that the equation may be satisfied.

INTENSITY OF SOUND WAVES

The intensity of sound waves is proportional to the amount of energy passing per second through unit area at right angles to the direction of propagation. Both kinetic and potential energy are present in a sound wave. The average kinetic energy equals the average potential energy, and the total energy at any time equals twice the average of either kinetic or potential, or the total energy equals the maximum of either kinetic or potential. Therefore the total energy of the sound wave may be determined by computing the maximum kinetic energy of all the molecules which are moving back and forth out of their equilibrium positions as the wave passes. If the sound wave is simple harmonic motion, the maximum velocity, u, of a vibrating particle of the transmitting medium is $2\pi aF$, where a is the amplitude, and F the frequency. The maximum kinetic energy of one particle, which also equals the

in phase when they continue to pass through corresponding points of their paths at the same time. For any other condition they are *out of phase*. They are in *phase opposition* when they reach their maximum displacement in opposite directions at the same instant.

The *wavelength* is the distance, measured along the direction of propagation, between two corresponding points of the wave train.

The general relation that exists among the frequency of vibration, the velocity of propagation, and the wavelength of wave motion in any medium is equally applicable to the propagation of sound waves in sea water. A body which is vibrating at a definite rate produces a disturbance that moves away as a wave in the surrounding medium. In the time, T, the vibrating body completes one vibration, and the wave advances a distance equal

total energy, E, of this particle, is

$$E=\tfrac{1}{2}mu^2=\tfrac{1}{2}m(2\pi aF)^2 \text{ ergs/particle.}$$

Let the density of the medium be ρ gm/cm^3. Then if the density, ρ, is substituted for the mass, m, the result is the *energy density* or the energy per unit of volume-

$$E=\tfrac{1}{2}\rho u^2=\tfrac{1}{2}\rho(2\pi aF)^2$$
$$E=2\pi^2\rho a^2F^2 \text{ ergs/cm}^3. \quad (1\text{-}2)$$

The loudness of a sound wave, which determines the strength of sensation, and its ease of reception depend upon the intensity, I, which is the *energy*

---

transmitted per second per unit area perpendicular to the direction of propagation. In 1 second the sound wave disturbs a volume of medium of length v, where v is the velocity of propagation. The intensity is therefore the energy in a column of medium of length v and unit cross section, the volume of which is v cm^3. Equation (1-2) for energy per unit volume or the energy density must be multiplied by v to find the intensity-

$$I=2\pi^2\rho a^2F^2v \text{ ergs/cm}^2\text{sec.} \quad (1\text{-}3)$$

From this result it is seen that the intensity of sound is proportional to the (1) square of the amplitude, (2) density of the medium, (3) velocity of propagation, and (4) square of the frequency of vibration.

Decibel System

The values of pressure, p, encountered in practice, range from about 10^{-4} to 10^6 dynes/cm^2. It is customary to express sound levels, L, in terms of the logarithm of sound-intensity ratios-

$$L=C\log(I/I_o) = C\log(p^2/p_o^2), \quad (1\text{-}4)$$

where C is a constant that depends upon the units used, and I_o is a specified value of sound intensity that is chosen as a standard of reference. In practice, C is taken as 10 and the corresponding unit for L is called the *decibel* (db). It follows that the pressure level of sound, or simply the sound level, L, is given by the equation,

$$L=10\log I = 20\log p \text{ db.} \quad (1\text{-}5)$$

A more practical concept of sound intensity is in terms of pressure variations which occur at all points in the transmitting medium as the sound wave advances. The greater the pressure variations, the more intense is the sound wave. It can be shown that the *intensity is proportional to the square of the pressure variation* at all frequencies. For most practical purposes, sound intensity in terms of pressure units is preferred to *energy density* or *energy flow* because of the ease of measuring sound pressure.

The logarithm is to the base 10.

Two reference units of pressure are in common use. These units are 1 dyne/cm^2 and 0.0002 dyne/cm^2. Both units have been used with underwater sound, but 1 dyne/cm^2 is used here, unless otherwise stated. In keeping with international practice, the unit 0.0002 dyne/cm^2 is used as a reference intensity for airborne sound.

If I exceeds I_o, L is positive, or the sound level is said to be "up" L db with respect to reference level I_o. If I is less than I_o, L is negative, or the sound level is said to be "down" L db with respect to reference level I_o.

Sound Propagation in an Ideal Medium

INVERSE SQUARE LAW

In a study of underwater sound it is important to understand how the sound intensity varies as the waves advance out from the source. Consider the most elementary condition-that of a very small radially pulsating sphere being placed in the medium. Its waves spread out spherically and affect the whole space occupied by the medium. If E (watts) is the total energy emitted from the source per second, the sound intensity I' at a concentric spherical surface of radius, r' (yards), is

$I' = E/(4\pi r'^2)$ watts/yd^2.

At any other concentric surface of radius r'' (yards), the sound intensity I'' is similarly expressed-

$I'' = E/(4\pi r''^2)$ watts/yd^2

or

$(I'/I'') = (r''^2/r'^2)$.   (1-6)

Thus the sound intensity at any surface varies inversely as the square of the distance of that surface from the sound source. This relation is commonly known as the *inverse square law*.

The inverse square law can be stated more simply by letting I be the intensity at any range, r, and I_1 the intensity at unit range (source intensity). Equation (1-6) becomes

$I = I_1/r^2$. (1-7)

Equation (1-7) expressed in the decibel system becomes

$L = L_1 - 20 \log r$, (1-8)

3

---

where "$L = 10 \log I$" is the sound level (db) at range r and "$L_1 = 10 \log I_1$" is the sound level at unit range. The quantity, L_1, is called the *source level*.

Because graphic presentation of data is often necessary for the interpretation of the principle, it is helpful to become familiar with the appearance of the foregoing equation plotted in different ways. The inverse square law, as expressed in equations (1-7) and (1-8), can be presented graphically in various ways.

In figure 1-1 the abscissa is proportional to *r*,

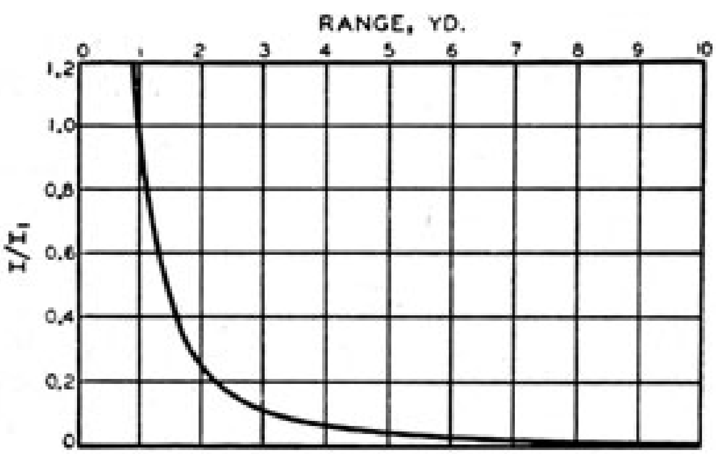


Figure 1-1. - I/I_1 as a function of range.

and the ordinate is I/I_1 or $1/r^2$. This method of presentation is not useful because the graph approaches too close to the horizontal axis to be visible beyond about 10 yards.

This objection is overcome by plotting

$$L - L_1 = 10 \log (I/I_1) = -20 \log r \quad (1\text{-}9)$$

as ordinate against *r* as abscissa. Such a graph is

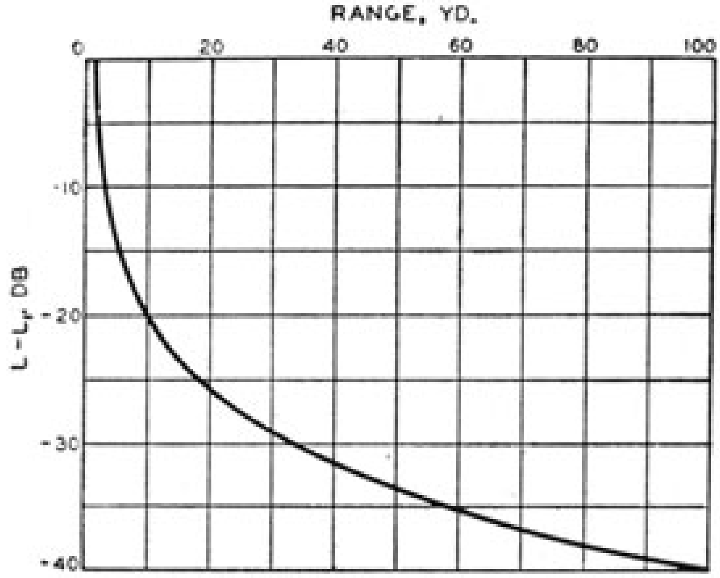


Figure 1-2. - L-L_1 as a function of range.

source of sound. It must be recognized that the character of the sound field is altered materially with a departure from a point source. In the ocean, the sound sources that come into consideration differ widely. Hulls of ships are large sources, emitting noise with a complicated spectrum. Sound projectors are moderately large sources, emitting relatively pure tones or sound of controlled frequency bands. Very small bubbles of air may become secondary sources of sound.

Any source can be considered to be divided into elementary areas, each of which acts as a point source of sound. If the linear dimensions of the source are small compared to the wavelength of the sound, the differences in the distances from a remote point in the sound field to any two elementary areas on the surface are small compared to the wavelength. Thus waves from the two elementary areas arrive at the remote point substantially at the same time. Under this condition the waves from the elementary areas add. The sound, moreover,

shown in figure 1-2. The expansion of the scale for small values of I/I_1 into large negative values of L-L_1, makes such a graph useful over a wider interval of ranges.

A third type of graph also uses L-L_1 as the ordinate but uses log r instead of r as abscissa. Figure 1-3 is the graph of equation (1-8) plotted in this way. This graph has two advantages-a much greater interval of ranges can be presented, and the graph of the inverse square law is a straight line.

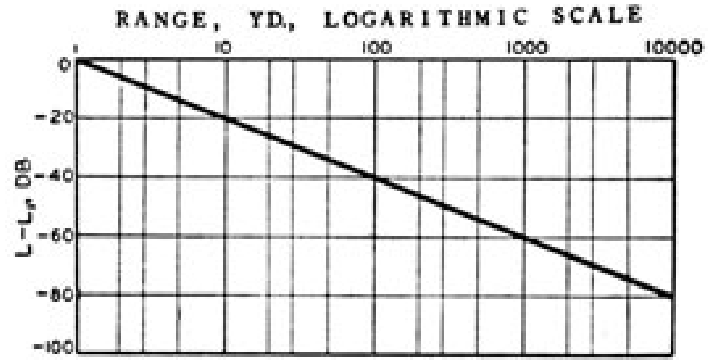


Figure 1-3. - L-L_1 as a function of range with the range being plotted on a logarithmic scale.

The foregoing discussion of the inverse square law was based upon the assumption of a point

is radiated uniformly in all directions. Under this condition the source can be called *small*.

If a source of simple harmonic waves is large compared to 1 wavelength, the waves do not arrive at a given point at the same time. Hence, there are *interference* effects, and the intensity radiated in some directions is greater than that radiated in others. It will be shown later that these interference effects are the basis for directional projectors.

If the source is large, for example a ship, and emits noise rather than single-frequency sound, the more obvious interference effects disappear. The intensity radiated in some directions, however, is still different from that radiated in others.

INFLUENCE OF DIRECTIVITY

A problem of basic consideration in sonar is the control of the distribution of sound energy radiating from a source. The reader is now

$b(\theta)=I_1(\theta)/I_a$. (1-10)

Then equation (1-7) becomes

$I=I_1(\theta)/r^2$

$I=I_a b(\theta)/r^2$ (1-11)

In converting to the logarithmic form, let L_a be 10 log I_a. L_a is called the *axial source level*. Because $b(\theta)$ is usually a proper fraction, its logarithm is negative and represents a reduction in sound level. To avoid confusion in use of signs, it is better to express this reduction as a positive number and subtract it than to add it as a negative number. It is therefore defined as B=-10 log $b(\theta)$. Thus equation (1-11) converted to logarithmic form becomes

$L=L_a$-B-20 log r db. (1-12)

The quantity, B, is called the *beam pattern*, or

familiar with the inverse square law and the general deviations from it when the sound source is not small. If the sound energy emitted by a source is confined to a cone or beam of small angle, the intensity is greater at a given distance than it would be in the case of a point source radiating uniformly in all directions. Such concentration of sound energy within a narrow beam is called *directional transmission*.

Equation (1-8) which gives the sound level in any direction at a range, r, may be altered to give the level in a directional sound field. This alteration could be accomplished by assigning a different value to the source intensity, I_1, for each direction; however, a simpler procedure is to designate the intensity at 1 yard in an arbitrary direction as the source intensity. The intensity in any other direction can then be obtained by multiplying by an appropriate factor determined by the direction. In the case of a sound projector that concentrates most of its energy in a beam, the value of the intensity at 1 yard from the source in the direction of the axis of the beam is considered to be the source intensity. Let the source intensity be I_a and let the intensity at 1 yard from the source in a direction making an angle θ with the axis of the beam be equal to $I_1(\theta)$ (read "I_1 of θ"), and the ratio of $I_1(\theta)$ to I_a be equal to $b(\theta)$. Thus-

directivity function. If equation (1-10) is expressed in logarithmic form, B is defined by

$$B = L_a - L_1 \quad (1\text{-}13)$$

At all points on the axis, B=0, because $b(\theta)$ is unity. Under this condition, equation (1-12) becomes equation (1-8).

Figures 1-4 and 1-5 show polar graphs of the function $b(\theta)$ and B for the same projector. These graphs have been calculated theoretically for a vibrating rectangular plate, the side of which is about 4 wavelengths long.

The graph of $b(\theta)$ (figure 1-4) shows that most of the sound is projected in directions which make

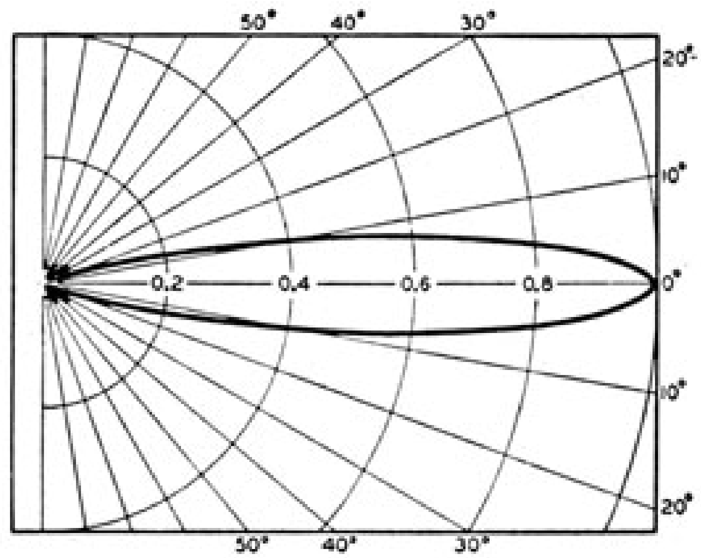


Figure 1-4. -Beam pattern of a projector.

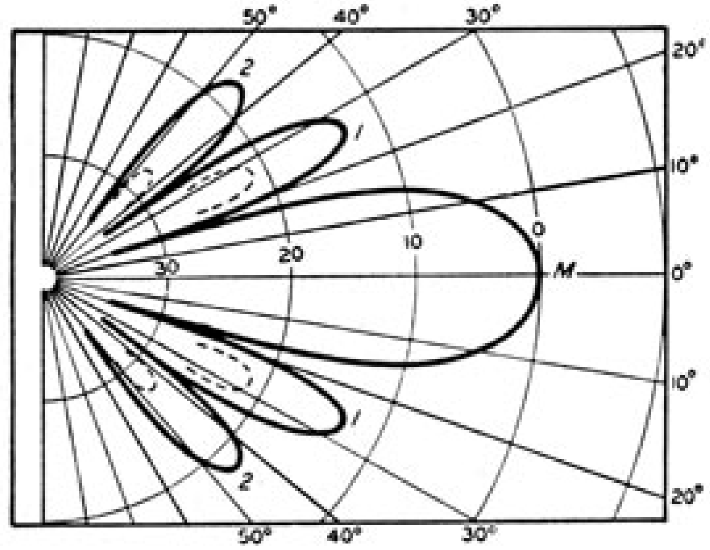


Figure 1-5. -Beam pattern of figure 1-4, with the ratio of ($I_1(\theta)/I_a$) expressed in decibels.

angles of less than 10° with the perpendicular to the plate. The radii represent the value of $b(\theta)$. Note the difficulty of showing the side lobes.

However, the very weak radiation at greater angles is important in some cases. Consequently, the graph of B (figure 1-5) is useful, because the logarithmic scale emphasizes these small intensities.

In figure 1-5 the maxima *M*, *1*, *2*, and others not shown are called *lobes*. *M* is the main lobe. For sonar bearings to be accurate, the main lobe should be narrow. The side lobes, *1* and *2*, are detrimental for many purposes, and in the design of modern projectors side lobe suppression is an important consideration. The dotted curves show the result of lobe suppression. With modern designs, the maxima of all side lobes are usually more than 20 db below the maximum of the main lobe. Graphs like those in figures 1-4 and 1-5 are drawn for projectors from actual measurements of sound level in different directions. They are called *directivity patterns*. Projectors and directivity are discussed in more detail in chapter 2.

DEVIATION FROM INVERSE SQUARE

The ocean, taken as a medium for the transmission of sound, is far different from the ideal condition assumed previously. The extent of the ocean is limited, being bounded by the surface and the bottom. It is not homogeneous-the upper layers are usually warmer than the lower ones and near the mouths of large rivers the salinity is greatly reduced. Because of both these facts the water is less dense in the upper layers. The temperature and salinity may change also in a horizontal direction. The pressure increases with depth. These changes in the physical character of the ocean cause variations in the velocity of sound waves being transmitted in the ocean.

Other less obvious acoustic properties of the ocean contribute to making the calculation of sound intensity difficult. As a sound wave travels outward from a source in the sea, some of the energy is converted into heat by friction because of the viscosity of the water. This process is called *absorption*. Another portion of the energy goes into the production of secondary wavelets which travel in directions other than that of the primary wave. This phenomenon is called *scattering*. A more general term, embracing both absorption and scattering, is *attenuation*.

It is possible to measure the total transmission loss and to observe how it deviates from the inverse square law value of the ideal medium. To measure transmission loss the axial source level, L_a, of the transmitting ship is kept constant and sound level L is measured at the receiving ship.

The difference

$$H = L_a - L \quad (1\text{-}14)$$

is the loss in level suffered by the sound in being transmitted from one ship to the other and is usually called the *transmission loss*. Except, for sign and the

LAW

The method by which the inverse square law is modified to represent the intensity or level of a sound field when the source does not radiate uniformly in all directions, has just been described. Deviation from the inverse square law when the sound transmission is in an actual medium such as the water of the ocean will now be considered.

effect of attenuation, this transmission loss is the same quantity as that plotted in figures 1-1 and 1-2.

Such an experiment shows that equation (1-12) does not accurately represent the actual transmission loss. The difference between the observed value of *H* and that calculated from equation (1-12) is thus a measure of the departure of the ocean from an ideal medium. This departure is often called the *transmission anomaly*.

It is sometimes difficult to isolate the effects of the beam pattern from other factors affecting transmission loss. Consequently, a more practicable

definition of transmission anomaly is the difference between the observed transmission loss and the transmission calculated from the inverse square law alone, without taking into account the directivity effect. The directivity effect is thus included in the transmission anomaly defined by

$A = H - 20 \log r$ (1-15)

whence the actual sound level can be calculated from the equation

$L = L_a - A - 20 \log r$. (1-16)

The usefulness of this concept of transmission anomaly is illustrated by figures 1-6 and 1-7.

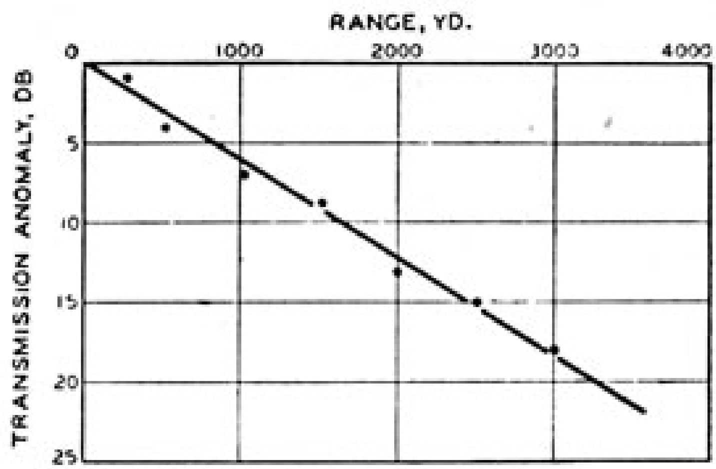


Figure 1-7. -The same experimental date as in figure 1-6, plotted as transmission anomaly.

range increases, is immediately apparent. Furthermore, a simple law is also obvious-the transmission anomaly, *A*, is proportional to range. Under favorable conditions, the transmission anomaly can be represented by the simple equation

$A = ar$ (1-17)

where *a* is an empirical constant called the *attenuation coefficient*.

Defined in this way, the transmission anomaly

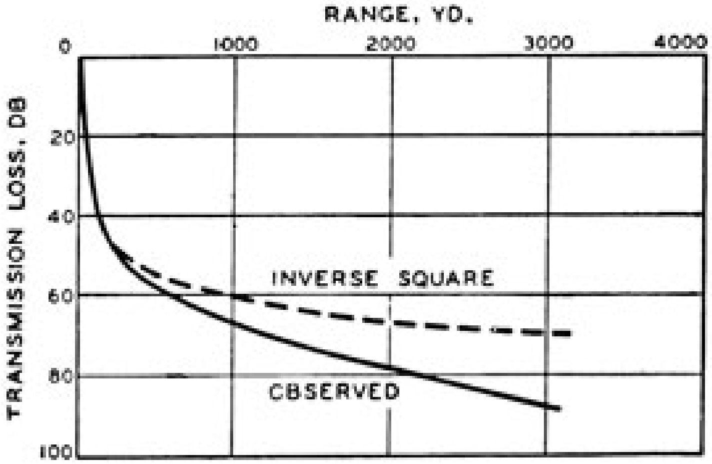


Figure 1-6. -Comparison of transmission loss observed in an experiment with that calculated from the inverse square law.

These figures are based on experimental data obtained under special conditions.

The solid curve of figure 1-6 is a graph of observed transmission loss *H* and, for comparison, the transmission loss calculated from the inverse square law is plotted as a dotted curve. The difference between these two curves does not seem very great, and would hardly be noticed if the dotted curve were omitted; yet the difference is very important in echo ranging.

Suppose the echo from a certain submarine can just be detected by a certain sonar equipment when the transmission loss is 70 db. If the inverse square law were valid, it could be detected out to 3,000 yards, but under actual transmission conditions it could not be detected beyond 1,250 yards, unless some other factor happened to be especially favorable at moderate ranges.

In figure 1-7 the increasing departure of the transmission loss from the inverse square law, as measures the difference in the transmission loss of sound from an actual source in the ocean and the loss of sound transmitted to the same range by a small source in an ideal medium. Besides the effect of directivity, other components of the transmission anomaly are:

1. Sound energy is converted into heat because of the viscosity of the water. This process is called *absorption*.

2. Variation in temperature and salinity cause changes in density, which, as the hydrostatic pressure increases with depth, result in variation of the velocity of the sound and consequent refraction of the sound rays.

3. The *scattering* of sound by reflection from the surface, by the bottom, and by particles suspended in the body of the ocean is a very important factor. A distinction should be made between *specular*, or *regular*, reflections-as from the surface and from the bottom-and the *diffused* reflections from the particles-ordinarily designated by the term "scattering."

4. Other factors about which little is known may contribute to the transmission anomaly.

Refraction of Sound

VELOCITY OF SOUND IN SEA WATER

In the foregoing discussion the refraction of sound in sea water was mentioned as an important factor in the transmission of sound in the ocean. In a homogeneous medium sound would travel in straight lines. As in the analogous case of light, the path of a sound wave is curved if the velocity of propagation is not the same at all points. A plane wave that enters another medium obliquely undergoes a change in direction, if the velocity of the wave in the second medium is different from that in the first. One part of the wave travels faster than the other and the wavefront is bent toward the medium of lower velocity. This phenomenon is called *refraction*. The ordinary laws of geometrical optics can be applied to the refraction of sound, although they are strictly true only for sounds of very high frequency, and do not take into account such phenomena as scattering, diffraction, reflection, and absorption. Although these phenomena cannot be ignored, it is simplest to omit them in initial discussions.

The velocity of sound in a liquid medium may be computed from the elasticity modulus, E, and density, ρ, of that medium-

v=square root(E/ρ). (1-18)

If E and p are in the British system of units, the velocity is in feet per second. As indicated by equation (1-18) the ratio of elasticity modulus to density of any transmitting medium determines the sound velocity in that medium. Any influence which changes either factor to give a change in the E/ρ ratio has a corresponding effect on the velocity.

The E/ρ ratio is governed by temperature, pressure, and salinity and the velocity must be evaluated for any given set of conditions. An increase in any of these factors will increase the sound velocity, although this increase is not directly proportional. Temperature, for example, ordinarily affects density to a greater degree than it affects the elasticity modulus. Thus, the higher the temperature of the medium the lower the density and the higher the velocity. Of the three factors (temperature, pressure, and salinity) that control the variables, E and ρ, in equation (1-18), temperature is by far the most important in sound transmission in sonar practice.

Note in figure 1-8 that the density changes at a variable rate with temperature. Thus, at constant salinity, the velocity increases with the temperature at a variable rate.

Figure 1-9 shows the variation with temperature for three salinities. Changes of 20° F, in the upper layer of the ocean are not uncommon. An increase in salinity of 1 part in 1,000 increases the velocity of sound 4.27 ft./sec. In most cases, however, the effect of salinity can be neglected

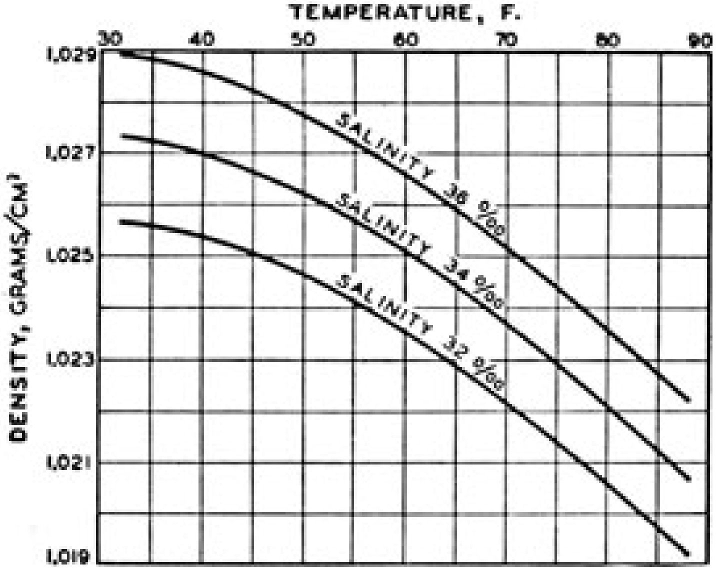


Figure 1-8. -Variation of the density of sea water with temperature and salinity.

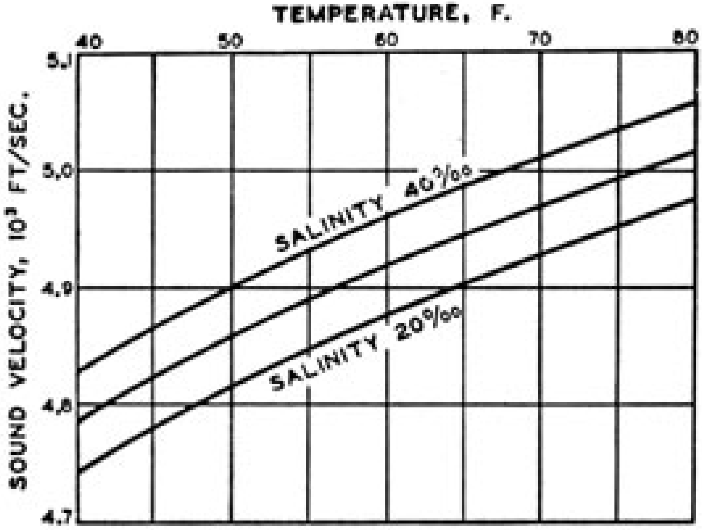


Figure 1-9. -Variation of the velocity of sound in sea water with temperature at three values of salinity.

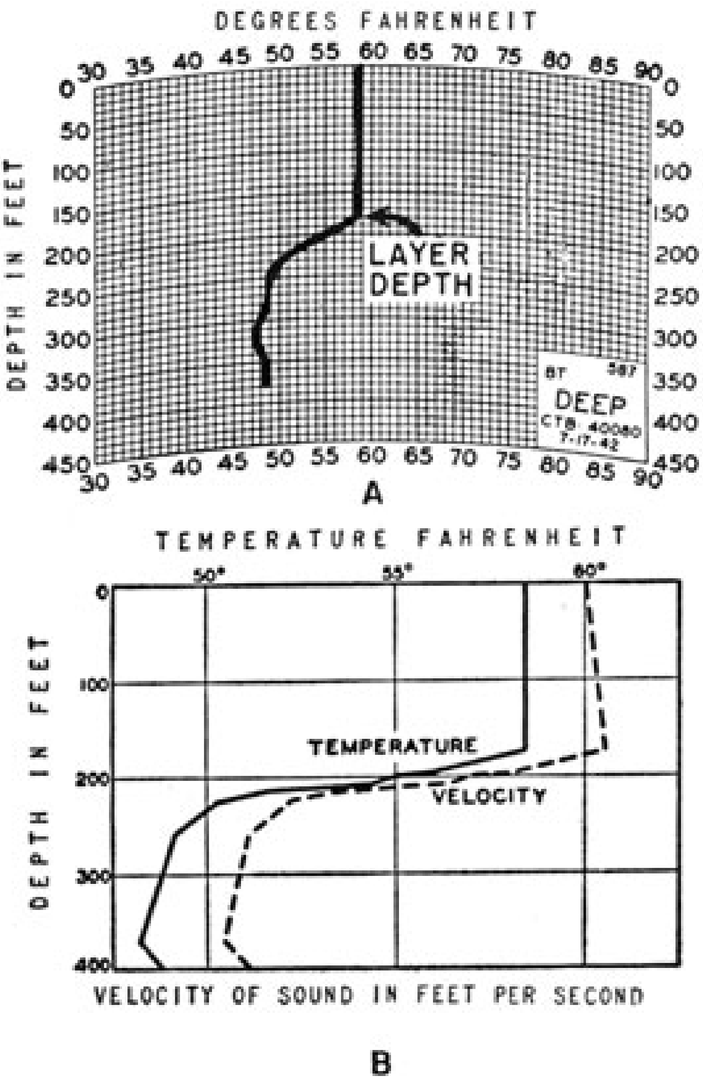


Figure 1-10. -Variation of temperature with

the velocity curve parallels the temperature curve quite closely.

At greater depths, temperature and salinity change only slightly, and the pressure effect dominates. The average temperature decreases with depth, as shown in figure 1-11, and down to about 2,500 feet, this decrease is sufficiently great to neutralize the effect of increasing salinity and pressure, so that the velocity of sound also decreases. At greater depths, the pressure effect begins to outweigh the temperature effect, and the sound velocity is seen to increase with depth. This minimum velocity at great depths has interesting acoustic consequences.

Horizontal and Vertical Changes

In considering temperature changes in the sea, it can be assumed that only variations in a vertical direction are significant. On this thesis the ocean may be considered as consisting of strata, in any one of which the same temperature exists over a large horizontal distance. Compared with vertical variations of temperature, the horizontal variations actually observed are very small. Changes in

depth. A, Typical slide; B, temperature-depth graph.

because salinity is comparatively constant except at the mouths of large rivers.

Increase of pressure with depth causes an increase in the speed of sound of 1.82 feet per second per 100 feet of depth. Figure 1-10, A, shows a typical bathythermograph slide with grid superimposed. The pressure effect is important only if both the temperature and the salinity are constant. This effect is shown in figure 1-10, B, in which the solid line shows how the temperature varies with depth in a particular case and the dotted line indicates the change in the velocity of sound with depth corresponding to this temperature distribution. The salinity effect is negligible. The effect of pressure on the velocity of sound in the isothermal layer of the upper 180 feet is evident from the velocity graph which shows a slight increase in the velocity with depth. Elsewhere temperature over a horizontal distance of 100 feet are rarely as much as 0.5° F and usually less than 0.1° F. Furthermore, they are not systematic

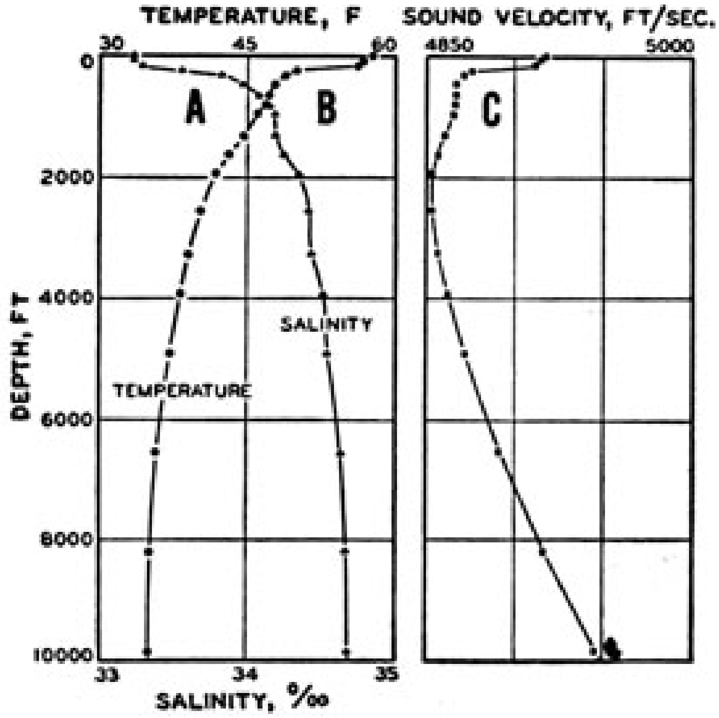


Figure 1-11. -Variation of temperature, salinity, and sound velocity with depth in the ocean.

On the other hand, over a vertical distance of 100 feet the temperature may vary as much as 10° F, as figure 1-10 shows.

It is now evident that temperature distribution with depth is the dominant factor in determining conditions for sound transmission in sea water. The *bathythermograph* was developed to determine this distribution. The bathythermograph is frequently referred to by the abbreviation BT. It is rugged and convenient in size, and can be lowered over the side for use while the vessel is underway. Furthermore, as it is lowered into the sea, the bathythermograph automatically draws a graph showing the temperature as a function of depth.

A functional schematic of the bathythermograph

is shown in figure 1-12. As the instrument is lowered, a stylus is moved by the thermal expansion

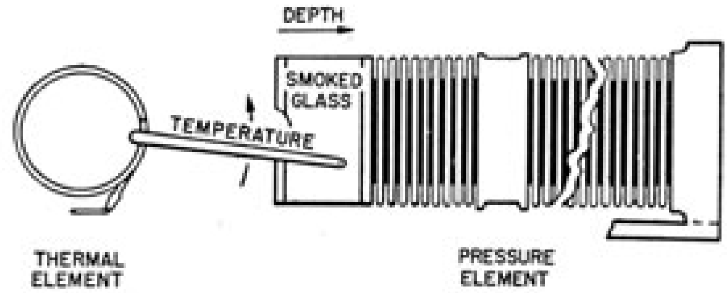


Figure 1-12. -Schematic of the bathythermograph.

or contraction of a liquid in the copper thermometer tube (thermal element). The increasing hydrostatic pressure compresses a bellows, which drives a smoked glass slide at right angles to the stylus which is driven by the thermal element. Thus a permanent graphical record of temperature against depth is obtained as the instrument is lowered and raised in the ocean. Figure 1-10, A, shows a typical slide with a coordinate grid superimposed; figure 1-10, B, is the temperature-depth graph made from the trace on the slide. Such temperature-depth graphs are called *bathythermograms*.

Twelve typical bathythermograms are shown in figure 1-13. These bathythermograms illustrate the variable character of the temperature distribution in the surface layers of the ocean. Examination of these charts shows that the temperature-depth curve can be subdivided usually into segments having different temperature gradients. Layers in which the temperature is uniform are called *isothermal layers* (figure 1-13, A). *Negative*

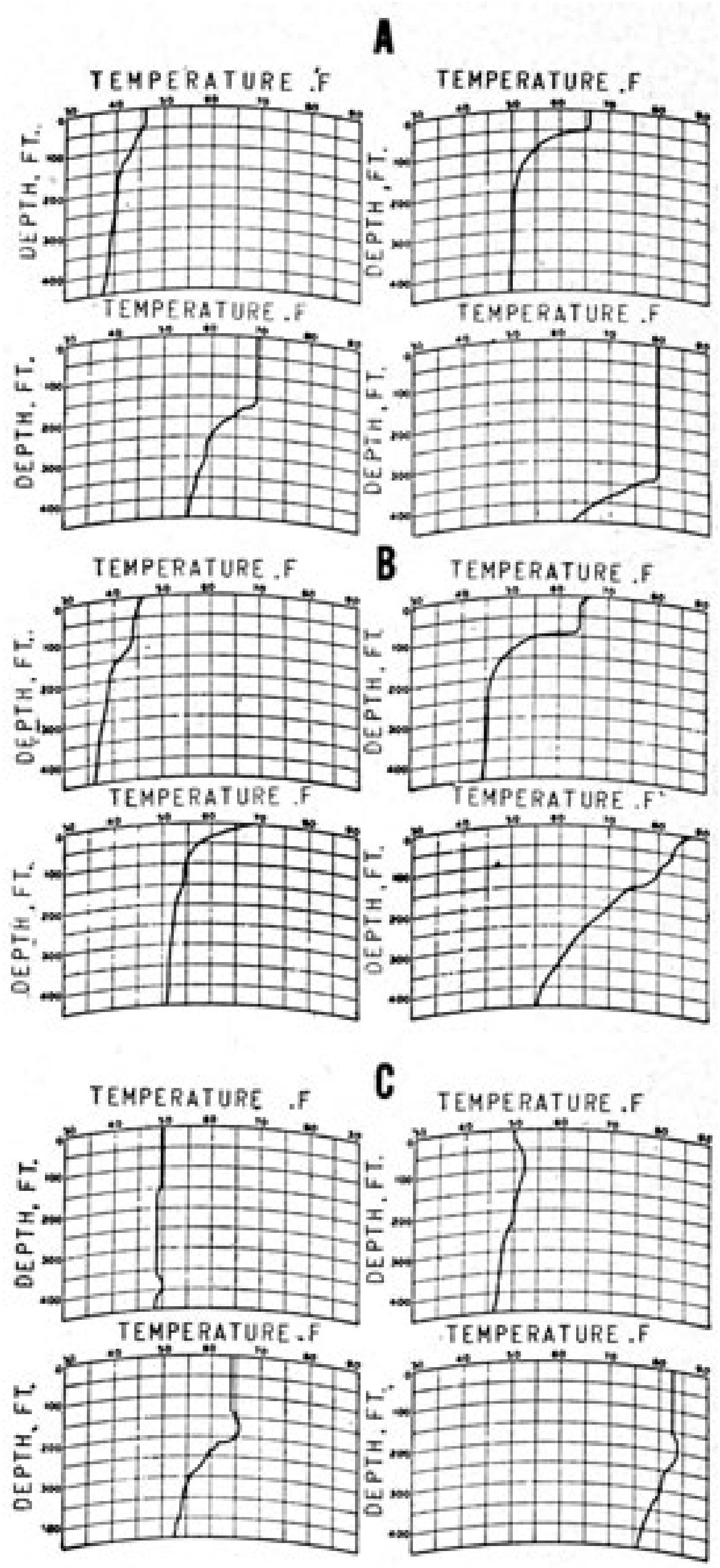


Figure 1-13. -Typical bathythermograms corresponding to various gradients. A, Isothermal surface layer; 13, negative temperature gradient in surface layer; C, positive temperature gradients.

gradients (figure 1-13, B) describe conditions in layers in which the temperature decreases with

depth. *Positive gradients* (figure 1-13, C) describe conditions in layers in which the temperature increases with depth.

A layer in which the temperature decreases very rapidly-particularly if it is immediately beneath an isothermal layer or a layer of smaller gradient-is commonly called a *thermocline*. The decrease in temperature which always occurs at great depth is sometimes called a *permanent thermocline*.

SNELL'S LAW OF REFRACTION

It has been pointed out how a sound beam is bent or curved from a straight path if it passes obliquely from one layer of sea water to a second layer where the velocity is different from that in the first layer. With a method of determining the velocity of sound at each point in the sea, it is theoretically possible to calculate the sound rays, or paths, along which the sound travels. If, for simplicity, the ocean is assumed to be stratified so that the temperature at all points having the same depth is the same, the calculation becomes quite simple.

No attempt is made here to give a detailed explanation of the computational methods. The computation is based on the familiar Snell's law of refraction that is discussed in all textbooks of physics as it applies to light rays. Figure 1-14 shows an especially simple case of three layers, or strata, in each of which the sound velocity is constant.

If a plane wave is considered to be passing through these three layers, Snell's law is

$$v_1/\cos(\theta_1) = v_2/\cos(\theta_2) = v_3/\cos(\theta_3) \quad (1\text{-}19)$$

where v_1 and θ_1 are the velocity and inclination of the ray in the first layer, and so on. Note that the angle of inclination, θ, is the complement of the angle usually given with Snell's law. The ray in each layer is a segment of a straight line; but if the layers are allowed to become very thin, the ray approaches a smooth curve. At each point along the ray, however, the relation between the inclination of the ray and the velocity of sound is still given by equation (1-19).

TYPICAL RAY DIAGRAMS

Because most velocity distributions can be approximated for series of layers from bathythermograms, an approximate ray construction can be carried out with the aid of Snell's law as indicated. Such a ray diagram represents the sound field produced by sound energy transmitted from a sonar projector. If an underwater target is located within the bounds of the ray diagram a return echo may be received at the sonar vessel.

Marked Downward Refraction

A ray diagram for typical conditions of sharp downward refraction is shown in figure 1-15. It should always be borne in mind that the curvature of the rays is greatly exaggerated because of the necessary contraction of the horizontal scale. In figure 1-15 the ratio of horizontal to vertical scale is 75 to 1.

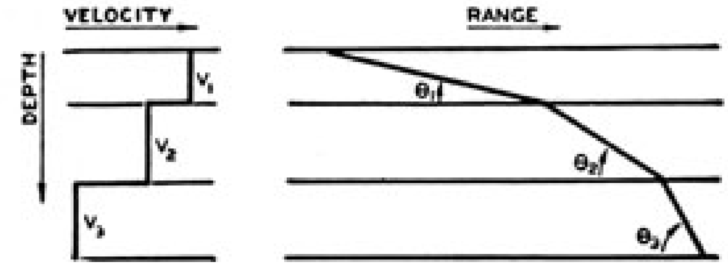


Figure 1-14. -Diagram illustrating Snell's law

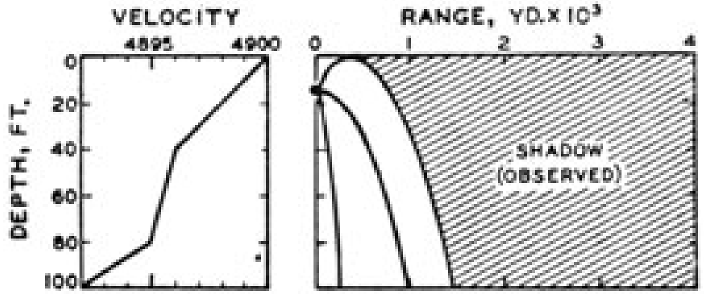


Figure 1-15. -Ray diagram with sharp downward refraction.

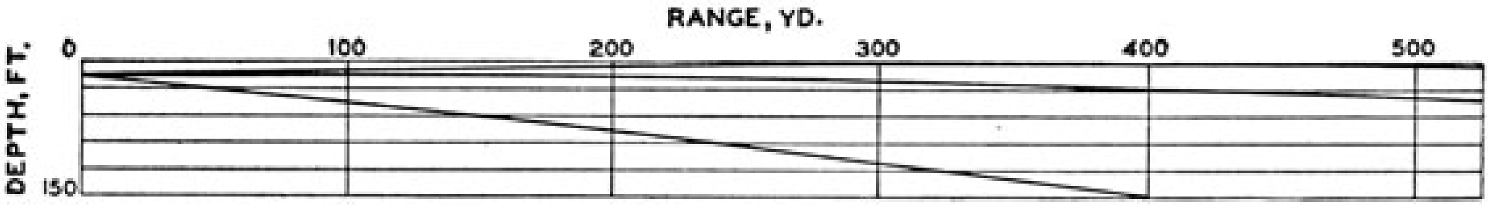


Figure 1-16. -Diagram of part of figure 1-15 drawn with undistorted scale.

Figure 1-16 shows a portion of the same diagram drawn on an undistorted scale.

The contracted horizontal scale also exaggerates the inclination of the rays with the horizontal. This inclination is shown in figure 1-17, the numbers being the true angles in degrees and the lines showing the angles as plotted on the diagram. The part of the beam above the axis is considered to have positive inclination; the part below the axis, negative inclination. In a directional transducer, nearly all of the energy is concentrated in a cone of about 10° opening. Hence a judicious selection of rays with initial inclinations of 5° or 6° on either side of the axis provides a sufficiently complete picture of the paths followed by the sound rays.

The velocity-depth graph of figure 1-15 shows three layers in which the velocity gradient is constant. The projector is at a depth of 16 feet. The following three rays are drawn:

1. The ray that leaves the projector at -6°, and which may be considered as the lower boundary of the main lobe of the projected beam of sound.

downward so that its inclination when it reaches the surface is zero. Any rays with inclinations greater than this critical value are reflected back by the surface inside the region bounded by the ray tangent to the surface.

A ray with less initial inclination does not reach the surface but curves down inside the critical ray; the 0° ray illustrates this point. The critical ray in the present example is the 1.4° ray. It bounds the direct sound field and for this reason is called the *limiting ray*.

Except for sound scattered or diffracted from the direct sound field, the shadow should be a region of silence. This picture is approximately a true one; observations made under conditions of strong downward refraction show a sharp drop of from 30 to 40 db in the sound level near the range indicated by the limiting ray.

The dimensions of the diagram do not permit the inclusion of the +6° (upper bounding) ray.

2. The ray that leaves the projector horizontally- the axial or 0° ray. This ray is shown bent sharply downward.

3. The ray that leaves the projector at +1.4°. This angle was chosen because this ray is tangent to the surface.

These three rays are also shown on figure 1-16 with an undistorted horizontal scale. The most striking feature of this ray diagram is that all the sound is confined to a very limited region and beyond about 500 yards from the projector the surface casts a shadow. The explanation of this shadow follows.

The outer rays of the upper half of the sound beam fall on the surface and are reflected there. A ray of a certain critical inclination is refracted

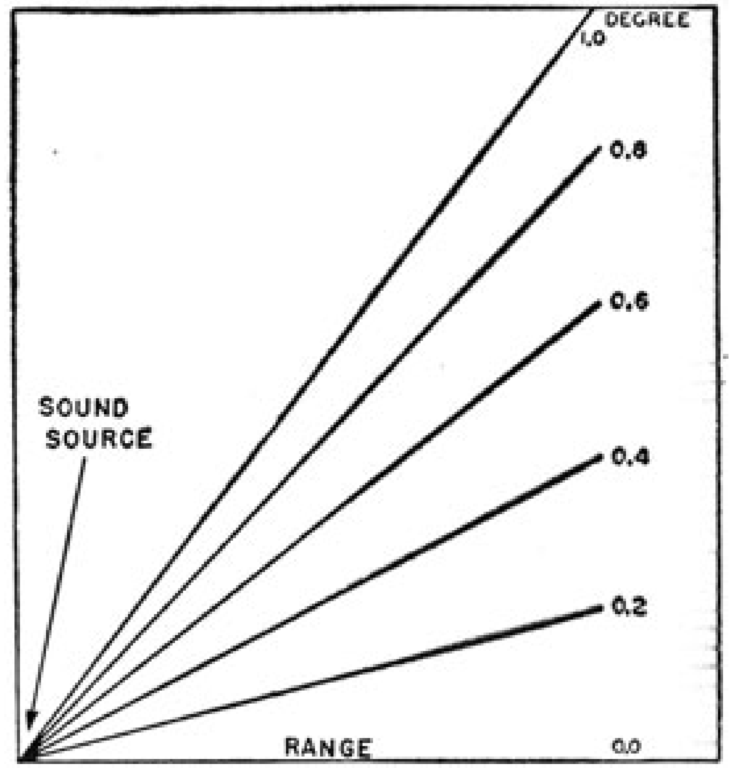


Figure 1-17. -Diagram showing how the inclination of the rays is distorted in the conventional ray diagram.

Isothermal Layer and Thermocline

Another common type of thermal distribution is shown in figure 1-10. This figure shows an isothermal layer at the surface, below which a sharp negative gradient occurs. In the isothermal layer, the velocity gradient is positive because of the pressure effect, as shown in figure 1-10, B. About 90 percent of the bathythermograph records taken all over the world show this type of thermal structure. The sound-velocity graph and ray diagram corresponding to this example are shown in figure 1-18.

a given depth decreases gradually with increasing range and shows no abrupt drop as the limiting ray is crossed. The intensity gradient is much greater below the "splitting" point than above.

Other thermal structures result in the sound field conditions illustrated by the ray diagrams in figures 1-19 and 1-20.

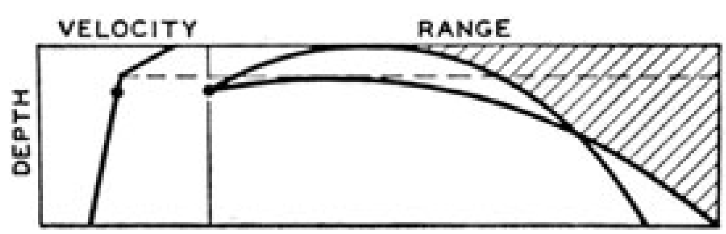


Figure 1-19. -Sound field bounded by two limiting rays.

Figure 1-19 illustrates the case in which two limiting

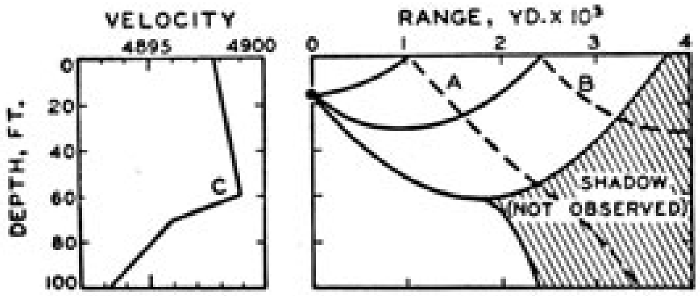


Figure 1-18. -Ray diagram for an isothermal surface layer.

Theory predicts a shadow, limited by the ray which is horizontal at the level of maximum velocity. The rays above the limiting rays are refracted upward and are ultimately reflected at the surface. Those below the limiting ray enter the thermocline and are there refracted downward. The sound beam is split along the limiting ray into an upper and a lower section; hence the term "split-beam pattern" is commonly applied to this type of ray diagram.

The shadow beyond the limiting ray might be expected to be a region of relative silence, as in the previous case. Actually the shadow in figure 1-18 differs from that in figure 1-16 in that it is penetrated by surface-reflected rays such as those designated by A and B. Because the surface reflects approximately all the incident sound energy, it is obvious that the shadow in figure 1-18 is not so complete as the one in figure 1-15. In the second velocity graph, the corner at the point of maximum velocity, C, is actually round instead of being sharp as shown. When this rounding is properly introduced to the diagram the "shadow" is found to be a region into which few rays, rather than none at all, penetrate.

Experiments show that there is no noticeable shadow under these conditions. The intensity at

229276°-53-2

rays bound the field.

Figure 1-20 shows a velocity distribution resulting in what is called a *sound channel*. All rays leaving the projector between rays A and B are alternately refracted up and down. The rays are thus confined to a certain layer, to which the term "sound channel" is applied. Transmission losses in sound channels are exceptionally low, and extremely long ranges are possible.

In the open sea, sound channels are rare and transitory in the upper layers, because the thermal conditions causing them are unstable. Near the mouths of large rivers, where salinity conditions cause changes in sound velocity, it is possible to have stable sound channels in the surface layers.

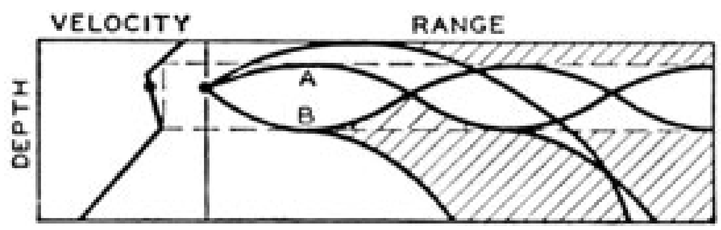


Figure 1-20 -Formation of a sound channel.

At great depths, where the temperature is practically constant, the pressure effect causes the sound velocity to increase with depth and there is a permanent sound channel. The extremely long ranges that are possible with low-frequency sound signals in this permanent sound channel are utilized in a long-range position-fixing system that uses signals from explosions set off at the depth of the sound channel. A full description of this system is given in chapter 16.

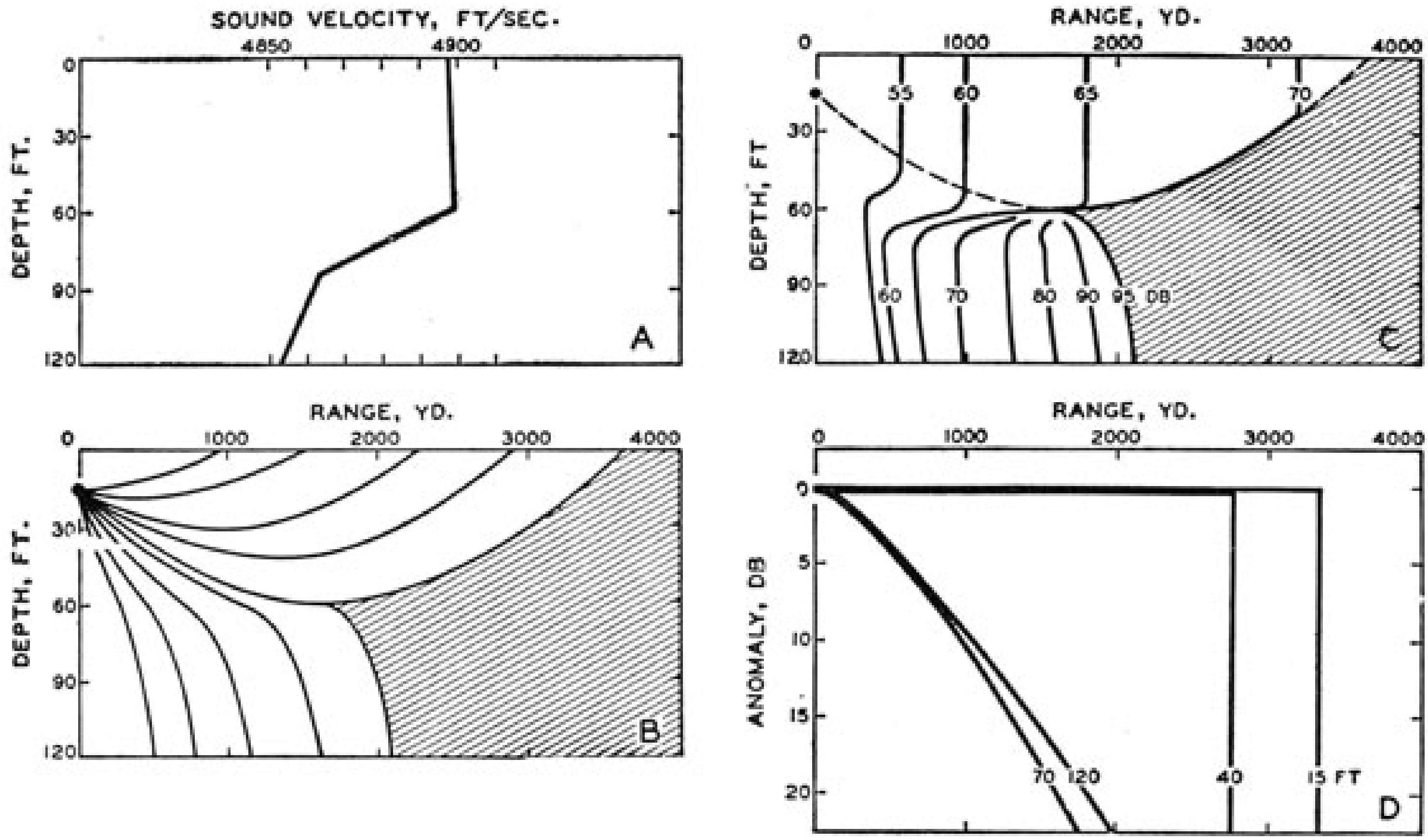


Figure 1-21 -Calculation of theoretical intensities for typical ray diagram. A, Bathythermogram; B, ray diagram; C, intensity contours; D, anomaly graph for several depths.

Ray Divergence

The effects of refraction have been presented in black-and-white pictures of silent shadows and regions of direct or reflected sound. This concept comes from the earliest form of theory on which echo-range predictions were based. However, it has since been found that the shadows are not silent and that there are marked variations within the field of direct sound.

Even before this experimental knowledge was obtained, attempts had been made to enlarge the ray theory to enable the calculation of intensity changes in the direct field. This intermediate theory is still useful for some purposes even though it also predicts completely silent shadows that are not observed.

values of transmission loss in db. Above the thermocline they represent the loss calculated from the inverse square law. In general, above the thermocline, these contours are farther from the projector than they are below the thermocline, and they are more widely spaced above than below. Throughout the whole shadow (shaded area) the calculated intensity is zero, and the transmission loss is consequently infinite.

Another method of presenting the results is shown by figure 1-21, D. The transmission anomaly was calculated for various points. If the depth is held constant-for example at 70 feet-and its distance from the source is allowed to vary, the series of values obtained can be plotted as a curve. See the curve marked "70 feet" in figure 1-21, D. These graphs are smooth curves when the depth is greater than that of the thermocline. When the point is above the thermocline, the transmission anomaly is practically

The results of these theoretical calculations can be presented graphically in several ways, as illustrated in figure 1-21. Figure 1-21, A, is a typical bathythermogram showing an isothermal layer and thermocline. The corresponding ray diagram is shown in figure 1-21, B. Figure 1-21, C, shows a series of contours on which the sound level is constant. These contours are identified by the

zero until the point reaches the shadow zone where it suddenly becomes infinite. The discontinuous change in the transmission anomaly is due partly to the approximate velocity-depth curve used in the

---

calculation. If these approximations were eliminated from the calculation, the change at the shadow would not be so abrupt.

The very marked increase in the transmission anomaly in the thermocline has important operational implications. From figure 1-21, C, it appears that if, for example, at a range of 1,000 yards a hydrophone is lowered to a depth of from 80 to 90 feet, it enters a region where the sound transmission is poorer by nearly 10 db than it is at from 20 to 30 feet higher. The sudden increase of the transmission anomaly is called the *layer effect*. The importance of the layer effect is enhanced by the prevalence of this type of thermal pattern in the ocean all over the world.

Figure 1-22 shows corresponding diagrams for a case of downward refraction.

INADEQUACY OF THE RAY THEORY

There has been much speculation about the reasons for the differences between the ray diagram theory and experiment-that is, the absence of sharp silence shadows and the presence of marked variations of intensity within the field of direct sound.

The failure to observe the sharply bounded silent shadow predicted by the ray-diagram theory should not be surprising. It is well known that even in the case of light, shadow boundaries are not sharp. The encroachment of a wave motion into the geometric shadow of an obstacle is known as *diffraction*.

Calculations of theoretical intensities have been made of sound fields for various sound waves around corners in air. As explained in textbooks on physics these diffraction effects increase with the wavelength of the wave disturbance, so that the ray theory becomes less and less correct as the wavelength increases. The wavelength of 24-kc sound in sea water is several inches and much longer than the wavelength of light, so that considerable diffraction of sound may be expected. Calculations have been made which show that the predicted effect due to diffraction is large enough to explain some of the irregularities in the transmission anomaly. However, the quantitative agreement between these diffraction calculations and measurements are not exact.

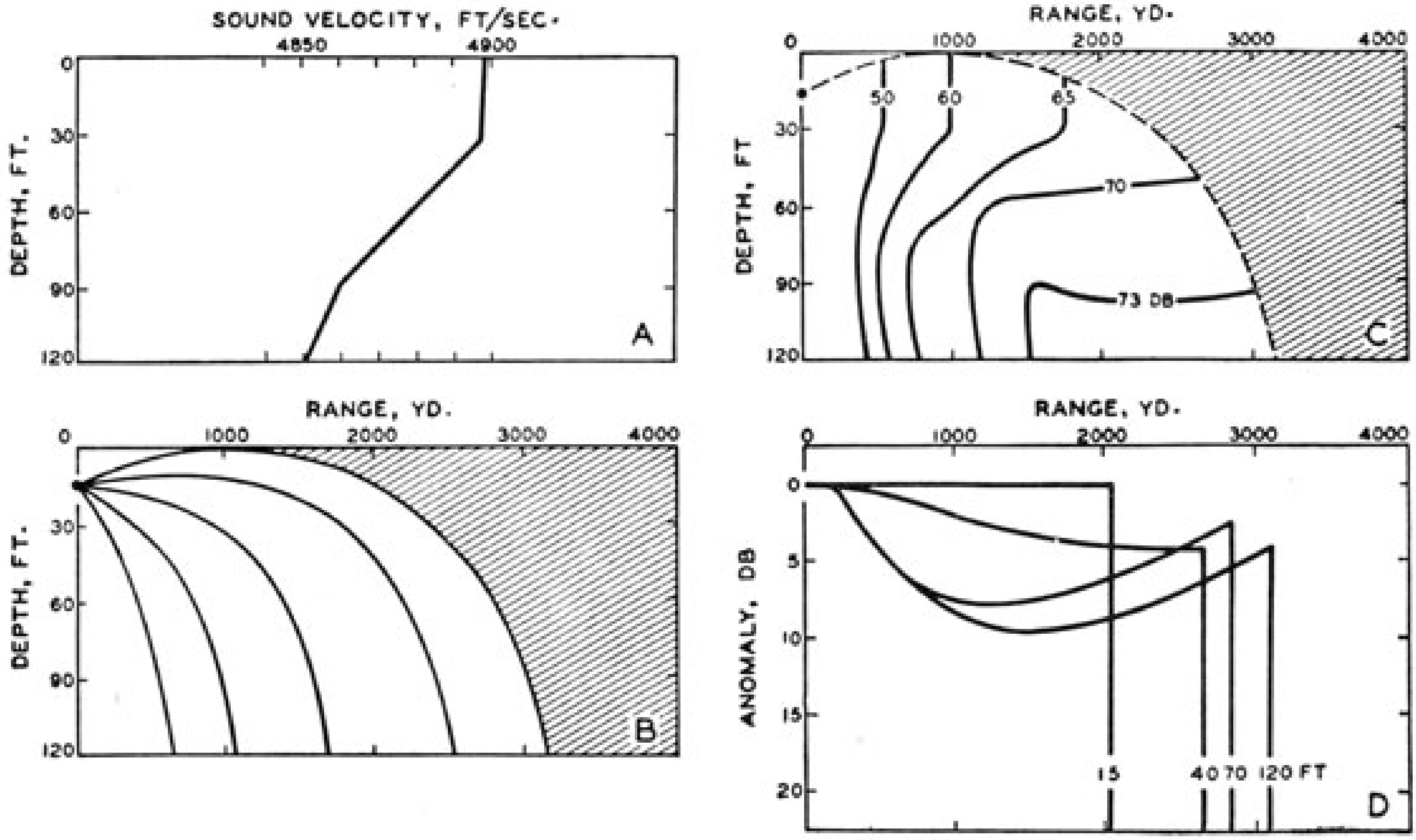


Figure 1-22. - Calculation of theoretical intensities for downward refraction.

Another possible explanation of the sound energy observed in the shadow is the scattering by small obstacles and particles suspended in the sea. The scattering of light by particles such as dust, snow flakes, and rain drops in the atmosphere is a familiar phenomenon and is known to be responsible for the many changes in the color of the sky and in the visibility of objects: The scattering of sound corresponding to this phenomenon occurs in the sea. For particles that are small compared with the wavelength, the relative amount of energy scattered depends surprisingly upon the wavelength.

This dependence is expressed quantitatively in *Rayleigh's law of scattering*: The relative amount of sound energy scattered by small particles in a medium is inversely proportional to the fourth power of the wavelength; or, qualitatively, the shorter the wavelength, the greater is the scattering. For example, the wavelength of 5-kc sound is 10 times that of 50-kc sound-that is, a small particle scatters 10,000 times more sound of 50-kc frequency than of 5-kc frequency. It is probable that scattering is the explanation for some of the variations in the ray theory.

Reflection and Scattering

The mechanism of scattering, with its resulting reverberation, and the mechanism of echo formation from underwater targets are very similar. They can be discussed conveniently at the same time.

When a short-tone pulse is sounded in a large, empty room, the sound echoes and re-echoes from the walls, ceiling, and floor for a considerable time. This phenomenon is called *reverberation*. It has been studied extensively by acoustic engineers, because it interferes with the understanding of speech and the enjoyment of music. A suitable wall covering deadens sound and eliminates reverberation.

When an echo-ranging pulse of sound is emitted into the ocean a phenomenon called reverberation is observed. Although the ocean has a floor and a ceiling, it lacks the four walls of a room, and neither the laws nor the causes of underwater reverberation should be confused with those of reverberation in acoustic engineering.

Theoretically, if the surface and bottom of the sea were mirror-flat and if there were no suspended matter (including fish) in the water, there would be no reverberation. Every departure from these ideal conditions results in an echo, usually a very weak echo. There are many irregularities on the ocean bottom, each wavelet on the surface and each suspended particle in the water probably contribute their individual echoes. The combined result is a scattering of sound in all directions. Some of this scattered sound comes back to the transducer and is heard in the sonar loudspeaker. This reverberation has very important connections with echo ranging.

Reverberation is therefore to be considered as the resultant of a large number of very weak echoes. Some of the targets producing these echoes are not very obvious, nor is much known concerning them. They may be air bubbles, suspended solid matter, organic matter such as plankton and the fish feeding on plankton, or minute inhomogeneities in the thermal structure. Minor irregularities of the ocean bed are very effective scatterers, and reverberation is very high when the sound beam strikes the bottom. The surface waves undoubtedly contribute appreciably to it.

Reverberation is easily distinguished from extraneous noise because reverberation is a tone of fairly definite pitch, whereas noise has a wide band of frequencies. The individual echoes mentioned as forming reverberations are not perceptible as such; they overlap one another in time, causing marked fluctuations in the intensity. If the signal is of constant frequency, transmitted horizontally, it is succeeded by a quavering, ringing tone of rapidly decreasing loudness, interspersed with occasional bursts of sound that might be mistaken for echoes by an inexperienced observer. In shallow water a crescendo, effect may be perceived after a certain interval because of sound that is scattered backward by the bottom.

If relatively long pings (transmissions of sound with a duration of about 200 milliseconds) of constant frequency are used, reverberation has a

musical sound. With shorter pings the musical character disappears; although the pitch can still be distinguished, the tone becomes rough and grating.

When a frequency-modulated signal is used, the reverberation loses its musical character. Some frequency modulation may occur because of improper functioning of the sonar oscillator. If the reverberation from long pings of supposedly constant frequency is not musical, the oscillator should be examined for frequency instability.

ECHO FORMATION

When a sound wave passes over an obstacle suspended in a medium, the medium is set into vibration and becomes a secondary source of sound. The amplitude of the vibration is proportional to the amplitude of the primary sound, and consequently the intensity of the secondary sound is also proportional to the intensity of the primary sound.

The simplest example is that of an object like a submarine or a large fish, with dimensions that are large compared to the wavelength of the sound. Such an object intercepts a certain amount of sound and casts an acoustic shadow. The intercepted power is reradiated as the secondary sound, or, as it is more usually called, the *echo*.

The amount of power intercepted is determined by the *target area* of the obstacle. For the present, the target area may be defined in a simplified manner by imagining a shadow cast by the obstacle to fall on a plane perpendicular to the sound rays. The shaded area is the target area, σ. In a sphere with a diameter, *d*, for example, it follows that the target area would be a circle of area

$$\sigma = \tfrac{1}{4}\pi d^2. \quad (1\text{-}20)$$

energy is reradiated as sound. Thus, the secondary sound power is

$$W_s = F\alpha\sigma. \quad (1\text{-}22)$$

The effect of absorption is thus the same as if the target area were reduced. This secondary sound is radiated in all directions, though not necessarily equally in all directions.

A sphere reradiates the sound equally in all directions and is thus the simplest example to treat. It may seem that the existence of a shadow is in contradiction to this statement; however, at great distances from the sphere, diffraction causes the shadow to disappear. Consequently, the statement is strictly correct only at a considerable distance from the spherical target.

At a great distance, *r*, the power, W_s, that is reradiated from the target flows through the whole area, $4\pi r^2$, of an imaginary spherical surface centered at the target. Hence, the energy flow of the secondary sound is

$$F_s = F\alpha\sigma / 4\pi r^2. \quad (1\text{-}23)$$

If the target is not spherical, it radiates more sound in some directions and less in others than is predicted by equation (1-23). But this equation nevertheless still is valid on the average. The target area already depends on the direction of the incident sound, and may also be considered to depend on the direction in which the sound is scattered and on the reflecting properties of the target. If target area is adjusted to account for these factors, an effective target area, σ', may be used in expressing the secondary energy in the field surrounding the target

$$F_s = F\sigma' / 4\pi r^2. \quad (1\text{-}24)$$

In irregular objects, the target area depends on the direction from which the sound is incident.

If F is the energy flow (in watts per unit area) at the obstacle and W is the total power intercepted,

$W=F\sigma$. (1-21)

If the target is perfectly reflecting, all this energy is reradiated as sound. If the target is not perfectly reflecting, only a fraction, α, of this

INTENSITY OF SCATTERED SOUND

Because the energy flow, F, is defined as the intensity, I, equation (1-24) may be written also as

$I_s=(I\sigma')/(4\pi r^2)$. (1-25)

Note that, in this equation, r is the distance from the target to the point at which the scattered intensity is being calculated. I_s represents the secondary intensity. The primary intensity itself,

I, depends on r', the distance from source to target, and in general r' does not equal r. Neglecting refraction, which has been implicit in all of the previous equations, the following equation is applicable:

$I=I_1 / (r')^2$.

Therefore,

$I_s=I_1\sigma' / (4\pi r^2(r')^2$. (1-26)

If the echo is received at the source of the sound as in practical echo ranging, $r=r'$ and hence

$I_s=I_1\sigma' / 4\pi r^2 r^4$.

The phenomenon of scattering or reverberation differs from echo formation only in that it results from the action of many relatively small targets rather than from one large target. The action of a single scatterer can still be described by equation (1-27).

The simplified definition of a target area fails completely when the scatterer has dimensions that

the bubble, d, and on the average pressure, p, of the gas in the bubble. The dependence on p arises because the compressibility of a gas depends on its pressure.

The sharpness of the resonance peak of the bubble is determined by a parameter, Q, analogous to that of electric circuits. The value of this parameter cannot be calculated readily but is certainly less than λ_R/d, where λ_R is the wavelength corresponding to F_R.

It is difficult to calculate the exact value of the effective cross section of an air bubble compressed in water. However, when excited by sound frequencies near resonance, the effective cross section or target area becomes very large and may approach λ_R. For example, at a depth of 66 feet where a bubble 0.02 inch in diameter has a resonant frequency of 20 kc, the target area may be several square inches.

For frequencies more than 1 octave below resonance, the target area is considerably less than the actual cross section and approximate calculations show that gas bubbles scatter low-frequency sound considerably more effectively than do solid particle s of the same size.

are less than the wavelength of the sound. The target area, or the *effective cross section*, of small solid or liquid particles is much less than their actual section in a ratio that is roughly $(\pi d/\lambda)^4$ where d is the diameter of the particle and λ is the wavelength of the sound.

There are occasions when air or vapor bubbles might be expected to exert an appreciable influence on the transmission of sound. It is difficult to understand how bubbles can exist permanently in the sea, because sea water is not saturated with air except very near the surface. There are several obvious sources of intermittent bubble formation: (1) Whitecaps; (2) the breaking of the bow wave, which causes bubbles to be washed under a ship and into its wake; and (3) the rotation of the propellers of ships or even submerged submarines.

An air bubble is much more compressible than the surrounding water. Under the influence of a sound wave, it therefore pulsates with a relatively large amplitude. If the pulsation is to be followed, the water immediately surrounding the bubble must oscillate with an amplitude considerably greater than that of the water at a distance. The mass of this surrounding water, coupled with the compressibility of the air, results in resonance at a frequency, F_R, which depends on the diameter of

The mathematical investigations on which the preceding discussion of air bubbles is based have been confined to spheres. Their extension to non-spherical objects is not simple, but has been carried out for some objects. It is clear that the same general laws govern the more general shapes. For example, a fish that is not too flat or elongated casts a shadow roughly equal in area to that of a sphere of the same volume.

Our ignorance of the reflection coefficient causes some uncertainty in these calculations. The reflection coefficient depends largely on the compressibility of the fish. If the fish has a swim bladder (air cavity), it probably is the most effective portion in reflecting sound. Similar principles apply to kelp and other forms of marine life. These plants have gas-filled floats and are therefore very good reflectors of sound.

The bottom is especially important in the production of reverberations. Such objects as boulders, pebbles, shells, and coral are all potential scatterers of sound. A smooth sand or mud bottom theoretically behaves more or less like a mirror and scatters little sound back to the source.

The waves on the sea surface also act like separate targets. The large surfaces reflect ultrasonic waves somewhat like curved mirrors. The effect of the smaller ripples is not clearly understood, but such ripples probably scatter the sound about equally in all directions.

THEORY OF REVERBERATION

None of the small scatterers just discussed returns an appreciable echo by itself. The simultaneous reception of the echoes from a large number of the scatterers constitutes what we call reverberation.

To understand the manner in which the scatterers cooperate in producing reverberation, consideration must be given to the manner in which a pulse of sound (a "ping") is propagated. If the duration of the pulse is t_o seconds, it consists of a train of waves the total length of which is vt_o, where v is the velocity of sound. This distance is called the *train* length of the pulse. Because v is 1,600 yd/sec, approximately, a pulse of duration 0.1 second (100 msec) results in a wave train 160 yards long. If the frequency is 24 kc, there are 24,000 X 0.1 = 2,400 complete waves in the train.

One-half the train length is called the *ping length*; a pulse lasting 0.1 second thus has a ping length of 80 yards. The ping length is a more useful concept than the train length, for two reasons.

In the first place, in echo ranging, the time required for the pulse to travel from projector to target and back to the receiver is measured. The clock is the range dial and is calibrated in terms of the range of the target that returned the echo- not in terms of time. If a target is at range r, the travel time is $2r/v$. Therefore, if the echo is a pulse of duration t_o, the range indication increases by the amount r_o=-$vt_o/2$ during the reception of the echo.

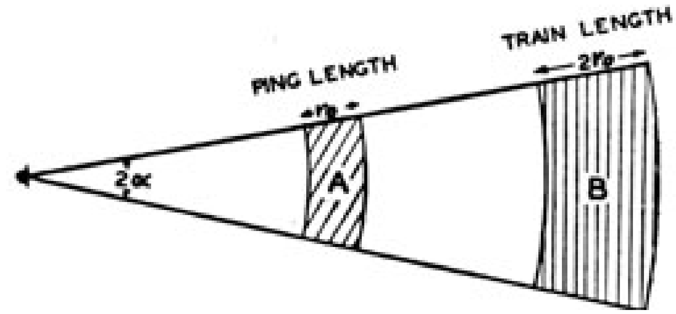


Figure 1-23 -Instantaneous relation between the region (A) from which echoes are being heard and the volume (B) occupied by the wave train for a beam whose angular half width is α radians.

instant, the actual train of waves no longer passes over this particular lot of scatterers; it has moved onward during the time the echoes were returning to the sonar. The instantaneous relation between volume *A* (from which the echoes are being heard), and volume *B* (which is occupied by the wave train), is shown in figure 1-23.

Figure 1-23 also shows graphically how the ping length and train length are related. Very little further reference is made to the train length, as almost no interest centers on region *B*. On the contrary, frequent reference to region *A* and the ping length is necessary.

The effect of scatterers suspended in the volume of the sea can now be calculated. Consider the simplest possible case:

1. There are *N* scatterers per unit volume.
2. Each scatterer has the effective target area σ'.
3. The sonar has a sharply defined beam of half width α. Its directivity pattern is shown in figure 1-24. The dotted line represents the axis of the beam.

This amount equals exactly the ping length as just defined.

In the second place, if there are many targets or scatterers, the echoes that are heard simultaneously come from those scatterers for which distance s from the sonar differ by less than r_o. At a given instant, therefore, echoes are received from all scatterers that lie in a spherical shell, with a thickness r_o, as shown in figure 1-23. At this

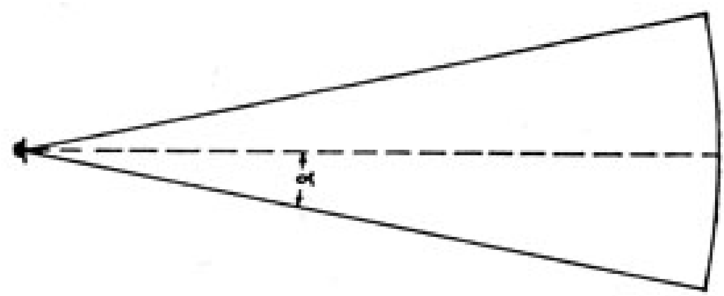


Figure 1-24 -Ideal beam pattern of half width α.

4. The sonar is in such a location that all effects of surface and bottom can be ignored.

The intensity of the echo from a single scatterer is given by equation (1-27), provided it is in the beam; otherwise, it is zero. There are many scatterers in the active shell (region *A*, figure 1-23) at

any instant. If V is the volume of this region, the number of scatterers whose echoes are being received is NV. If this number is combined with equation (1-27), the intensity of the reverberation is

$$I_r = I_1 N V \sigma' / 4\pi r^4 \quad (1\text{-}28)$$

Now the volume *V* is easily calculated. It is given approximately by

$$V = 2\pi r^2 r_o (1 - \cos\alpha), \quad (1\text{-}29)$$

where *r* is the range to the center of region A. Hence,

$$I_R = I_1 \left((N\sigma' r_o)(1 - \cos\alpha)\right) / 2r^2. \quad (1\text{-}30)$$

Several conclusions can be drawn from equation (1-30). A brief list of the simpler conclusions follows:

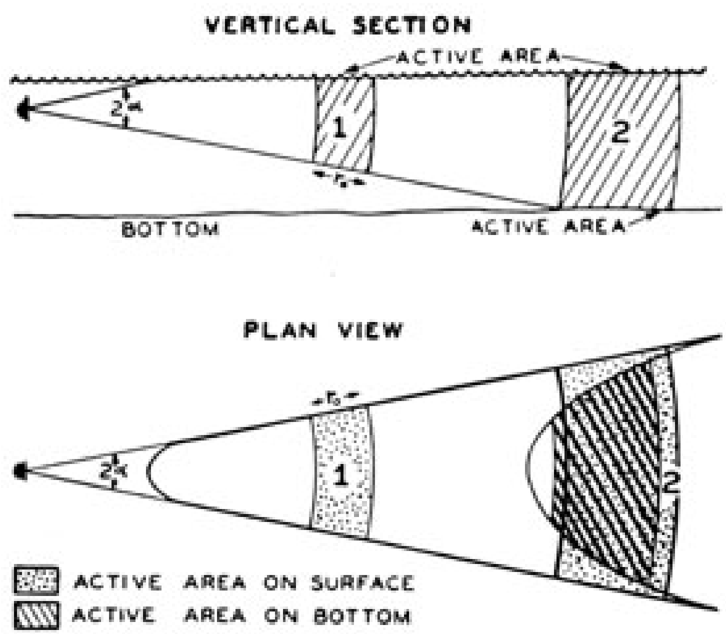


Figure 1-26 -Active areas on surface and bottom for two different positions of the wave train.

which shows that the echo from a single target varies inversely as the fourth power of *r*. The reason for the difference is the increase in the active volume, *V* (region *A*, figure 1-23), as *r* increases.

The theory of volume reverberation, as presented in the previous paragraph, requires only slight

1. Because the reverberation intensity, I_R, is proportional to the source intensity, I_1 increased sound output increases the reverberation.

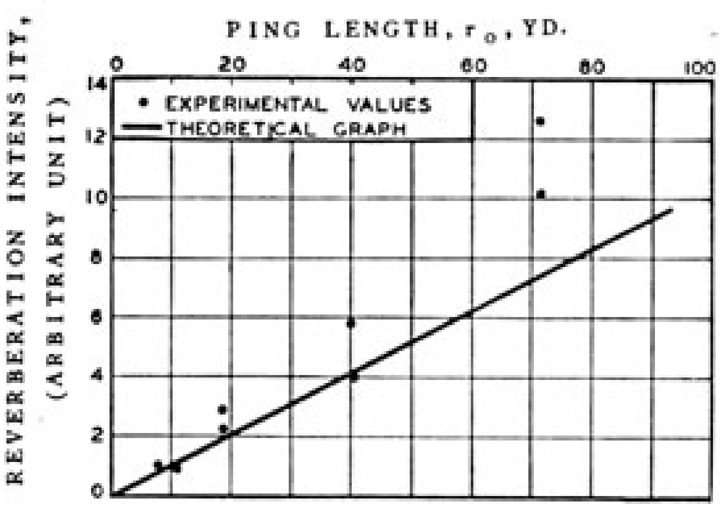


Figure 1-25 -Relation between ping length and reverberation intensity.

2. Because the reverberation intensity is proportional to the ping length, r_o, a long ping causes more reverberation than a short one. (See figure 1-25.) If the reverberation intensity were strictly proportional to the ping length, the dots would lie on the solid graph.

3. Because (1-cos α) increases as a increases, a broad beam causes more reverberation than a narrow one. In general, doubling the width of the beam causes I_R to increase about fourfold.

4. The (volume) reverberation intensity varies inversely as the square of the range, r; this relation should be compared with equation (1-27),

modification when the scatterers are located on either the surface or the bottom. These two cases are, in many ways, identical. Instead of an active volume, V, an active area, A, must be dealt with, namely, the area of the intersection of the surface (or bottom) with region A of figure 1-23, already discussed. In figure 1-26, which is similar to figure 1-23, two successive locations of active volume are shown. Until the beam intersects the bottom, there is no active area on the

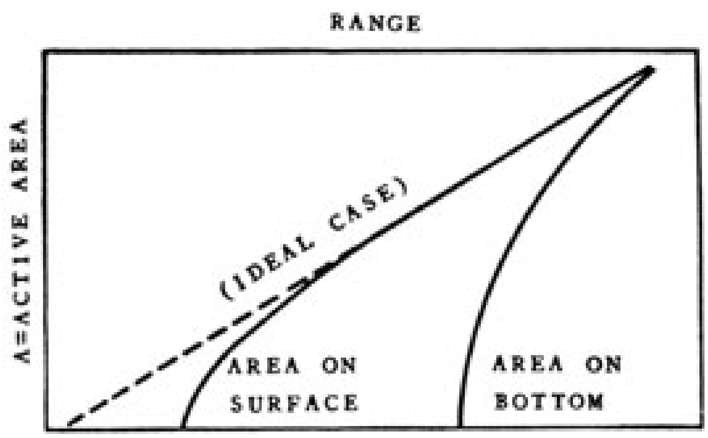


Figure 1-27 -Variation of active areas on surface and bottom as a function of range, when the projector is very close to the surface.

bottom; at position *1*, there is an active area on the surface, but none on the bottom. After some time, position *2* is reached and there is an active area on the bottom as well as on the surface. Figure 1-26 is drawn for a sonar mounted on a surface vessel; if the sonar were on a submarine near the bottom, the situation would be reversed. Note that at very short range there is no active area on either bottom or surface; this condition is shown in greater detail in figure 1-27.

The mathematical expression for the active areas is rather complicated, except in the special case in which the projector is very close to the surface. In such a case

$$A = 2\alpha r_0 r, \quad (1\text{-}31)$$

where α is to be expressed in radians. The graph of this equation is shown as a dotted line in figure 1-27. The departures at short ranges are obvious.

For simplicity it will be assumed that there are N' scatterers per unit of active area and that each scatterer has the target area σ'. The intensity of reverberation is (compare with equation 1-28)

$$I_R = (I_1 N' A \sigma') / (4\pi r^4). \quad (1\text{-}32)$$

If the range r is great enough so that equation (1-31) can be used for A,

$$I_R = (I_1 N' \sigma' r_0 \alpha) / (2\pi r^3). \quad (1\text{-}33)$$

Conclusions (1) and (2) drawn from equation (1-30) apply to equation (1-33) also. Conclusion (3) requires only slight modification, because (1-cos α) is replaced by α. Consequently, doubling the width of the beam increases surface reverberation by a factor of only two rather than

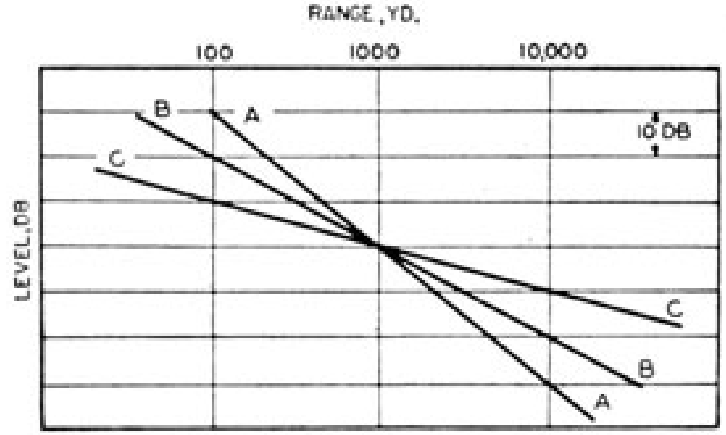


Figure 1-29 -Comparative levels of (A) echo from a single target, (B) surface (or bottom) reverberation, and (C) volume reverberation.

four. Finally, surface reverberation varies inversely as the third power of the range, while volume reverberation varies as the inverse second power of the range.

If the range is not great enough so that equation (1-31) can be used, somewhat more elaborate calculations are needed. The first three conclusions concerning volume reverberation apply without appreciable change, however, and only the dependence on range is changed. The graphs of figure 1-28 show this dependence on range for surface and bottom reverberations. In this figure it has been assumed that N', the number of scatterers per unit of active area, has the same value for both surface and bottom. Actually N' has a much greater value for the bottom than for the surface. This condition results in shifting the graph of bottom reverberation upward relative to the surface graph.

Figure 1-29 shows comparative levels of (1) an echo from a single target, (2) volume reverberation, and (3) surface (or bottom) reverberation, as calculated from equations (1-27, 1-30, and 1-33) respectively. To give a standard of comparison, it is assumed that all three factors have the same level at 1,000 yards, although this assumption is not necessarily the case in practice. Note that, at ranges of less than 1,000 yards, the levels *increase* in the following order: (1)

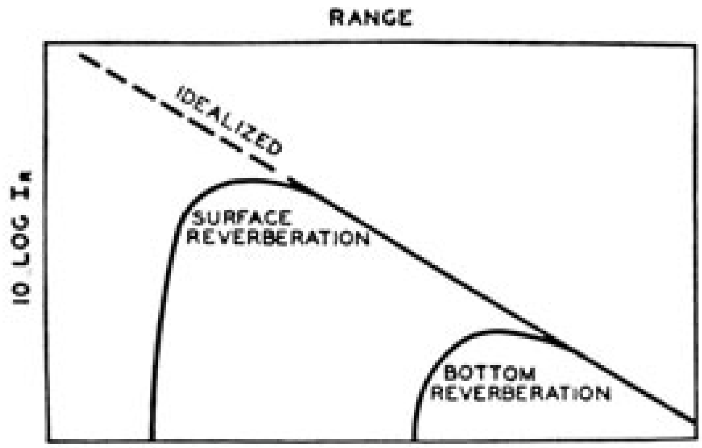


Figure 1-28 -Dependence of surface and bottom reverberation on range.

Volume reverberation, (2) surface (or bottom) reverberation, and (3) echo. At longer ranges they *decrease* in this same order. The graphs diverge 10 db from their neighbors for each tenfold increase or decrease in range.

Figure 1-30 -Oscillograms of reverberation and echo. Note that figure 1-29 does not show the dependence of surface and bottom reverberation upon range; to show this dependence it should be modified in accordance with figure 1-28.

REVERBERATION IN NONIDEAL CONDITIONS

All of the preceding calculations have been based on a number of simplifying assumptions that are not correct under actual conditions but are useful in presenting the basic ideas. The complications introduced by departures from the ideal cases just examined will now be considered.

The first simplification was that the scatterers all have the same target area, σ', and that there are *N* of them in each unit volume (or *N*' on each unit area). Obviously, the scatterers are not all the same, but because only the combination *N*σ' enters the final equation, this assumption does not cause any particular trouble. It is seen the $m=N\sigma'$ is the total target area of all the scatterers in a unit volume. This quantity is called the *volume-scattering coefficient*. Because *N* is measured in yd^{-3} and σ' in yd^2, *m* is measured in yd^{-1}; that is, 1/*m* is a length. It is essentially the distance a wave of the main lobe of the transducer. Let a be redefined as the angle (in degrees) at which the beam pattern has a value 6 db below the maximum (or axial) level. Then the values of K_s and K_v are given approximately by the equations,

$$K_s=4.2 \times 10^{-3}\alpha \quad (1\text{-}36)$$

and

$$K_v=4\pi K_s^2=5.5 \times 10^{-5}\alpha^2. \quad (1\text{-}37)$$

Note that the scattering coefficients are independent of the projector, whereas K_s, and K_v are independent of the ocean.

Finally, it has been assumed implicitly that the sound rays are straight lines and that the inverse square law determines the whole transmission loss. In actual cases the departures from ideal laws introduce marked effects, which can be ascribed to departures from the inverse square law of transmission loss.

In order to deal with these complications in as simple a manner as possible it is convenient to

train can travel before much of its energy is scattered.

In the same way, $n=N'\sigma'$ is the total target area of all the scatterers located on a unit area; it is called the *surface*- or *bottom-scattering coefficient*. Because N' is measured in yd^{-2} and σ' in yd^2, n is dimensionless; that is, it has the same numerical value whether yards or feet are used as units.

The second simplification is the assumption that the projector emits the sound in a sharply defined beam, with no side lobes. When actual projectors are involved, the factor (1-cos α) in equation (1-29) and the factor α in equation (1-31) must be replaced by others, the exact values of which depend on the beam patterns of the projector. If these factors are called K_v and K_s, respectively; equation (1-30) and (1-33) then become respectively

$I_R=(I_1K_v mr_o) / 2r^2$ (volume reverberation) (1-34)

and

$I_R=(I_1K_s nr_o) / 2\pi r^3$ (surface reverberation) (1-35)

The two factors, K_v and K_s, like the ones they replace, bear a simple relation to the half-width define the *reverberation level*, *RL*, by

$RL=10 \log(I_R/I_1)$ db. (1-38)

Note that *RL* is independent of the sound output of the sonar.

The *volume*- and *surface-reverberation indices*, J_v and J_s, are defined by

$J_v=10 \log K_v$ (1-39)

and

$J_s=10 \log K_s$ (1-40)

respectively and, with these substitutions, equations (1-34) and (1-35) become respectively

$RL_v=J_v+10 \log(mr_o/2)-20 \log r$ (volume) (1-41)

and

$RL_s=J_s+10 \log(nr_o/2\pi)-20 \log r$ (surface) (1-42)

Equations (1-41) and (1-42) are correct only if the transmission of sound is accurately given by the inverse square law. It can be shown that the departures from the inverse square law are in most cases properly taken into account in the following equations:

$RL_v=J_v+10 \log(mr_o/2)-2H_v+20 \log r$ (1-43)

and

$RL_s=J_s+10 \log(nr_o/2\pi)-2H_s+20 \log r$ (surface) (1-44)

where H_v and H_s are the actual transmission losses

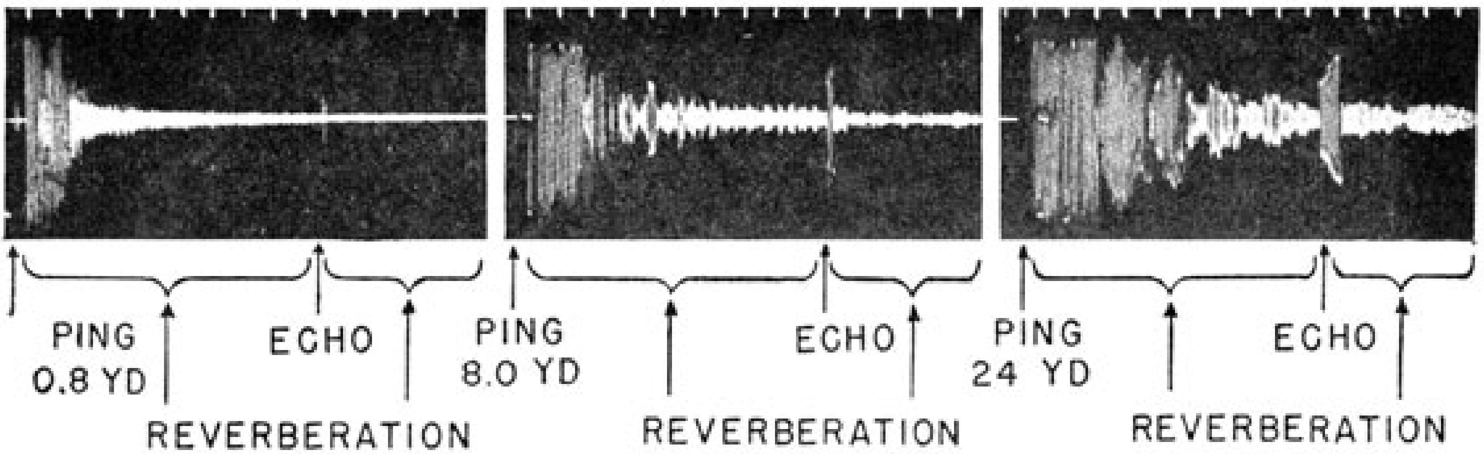


Figure 1-30. -Oscillograms of reverberation and echo.

from the sonar to the active regions responsible for the reverberation. It is easily seen that if H_v=H_s=20 log r, equations (1-43) and (1-44) reduce to equations (1-41) and (1-42).

The form of equations (1-41) and (1-42) suggests that the reverberation decreases steadily with time from an initial high level. This is not true. The ringing sound mentioned earlier in the discussion indicates that rapid changes in the intensity occur, that are not predicted by these equations. The oscillograms of recorded reverberation show these changes, as in figure 1-30.

These oscillograms are typical of the experimental data in this field and will be discussed in some detail. The three oscillograms were taken in rapid succession with different ping lengths of 0.8 yard, 8.0 yards, and 24 yards. Range marks are spaced 40 yards apart at the upper edge. The electric input to the transducer was coupled to the oscillograph and is recorded at the extreme left. This recording of the electric input is followed by a blank interval of about 0.025 second, during which the connections were changed from *send* to *receive*. The portions of the trace to the right of

The theory presented above asserts that the intensity of the reverberation should be proportional to the ping length, r_o. Consequently, the amplitudes of reverberation should be proportional to $r_o^{1/2}$ so that the three oscillograms should show amplitude of approximately 1:3.2:5.5. It is obviously difficult to verify this by a *single* measurement, because of the rapid and irregular fluctuations in the amplitude of the reverberation. On the average, these ratios are quite close.

A more detailed study of the problem shows that the theory described here refers only to such average values, and that there is a good explanation of the rapid changes in amplitude. Two possible causes immediately suggest themselves:

1. The number of scatterers in the active region varies as the active region moves outward.
2. The echoes from the different scatterers interfere.

The first of these possible causes is easily seen to bring about some fluctuation, but it is often relatively unimportant as compared to the second. If there are many small scatterers, only the second cause need be considered. As the number of

this interval are reverberation, except for the echo, which is clearly visible in each. The early reverberation is so intense that it is off scale in the two right-hand examples. The ordinates of the three oscillograms are comparable, except that the electric circuit for recording the outgoing ping did not respond fully to the very short 0.8-yard ping. The receiving circuits, however, responded fully to its echo. This echo is rather weak, but the other two echoes have the same amplitude.

scatterers in the active region decreases, the relative importance of the first cause increases.

Thus the second cause would dominate in long pings (large active regions) and the first cause would dominate in exceedingly short pings (small active regions). An inspection suggests, however, that even for the 0.8-yard oscillogram, the second cause of fluctuation is important, although some of the long "spines" may be caused by single scatterers.

REVERBERATION WITH A HORIZONTALLY DIRECTED BEAM

In the usual echo-ranging condition, the transducer is directed horizontally in deep water, and both surface and volume reverberation are generally observed. The intensity of the resulting reverberation at each range therefore depends on which of these two types of reverberation is dominant. Thus, as shown by the following explanation, volume reverberation always dominates at long ranges, whereas surface reverberation usually dominates at short ranges.

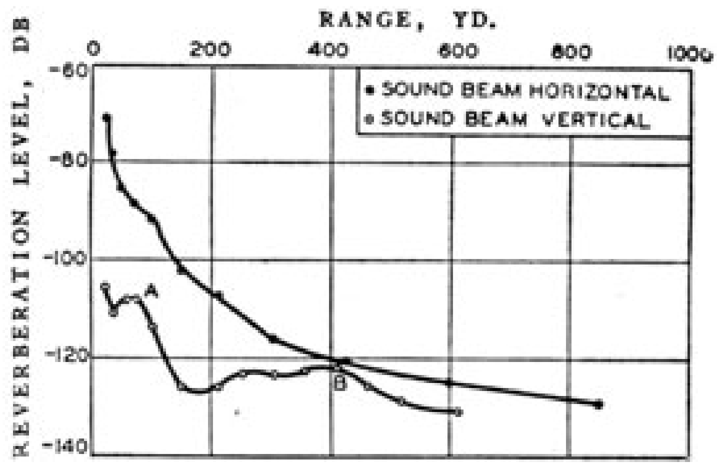


Figure 1-32 -Comparison of reverberation at wind speed of 17 miles per hour with horizontal and vertical beam.

It is convenient to begin the discussion with average reverberation-range curves obtained under practical echo-ranging conditions. Surface and volume reverberation then are considered separately in more detail; finally, average values of the scattering coefficients are given.

Two reverberation curves are shown in figure 1-31; they are averages of observed reverberations at high and low wind speeds. The measurements were made at 24 kc, using echo-ranging equipment with the transducer mounted at a depth of 16 feet and a standard ping length of 80 yards.

The two curves exhibit the following features:

1. At short ranges (less than 500 yards) the average reverberation level depends strongly on the roughness of the sea surface as measured by wind speed.
2. At long ranges (beyond 1,000 yards the average reverberation is independent of wind speed.
3. With high wind speed the reverberation level drops rapidly.
4. With low wind speeds, the reverberation drops

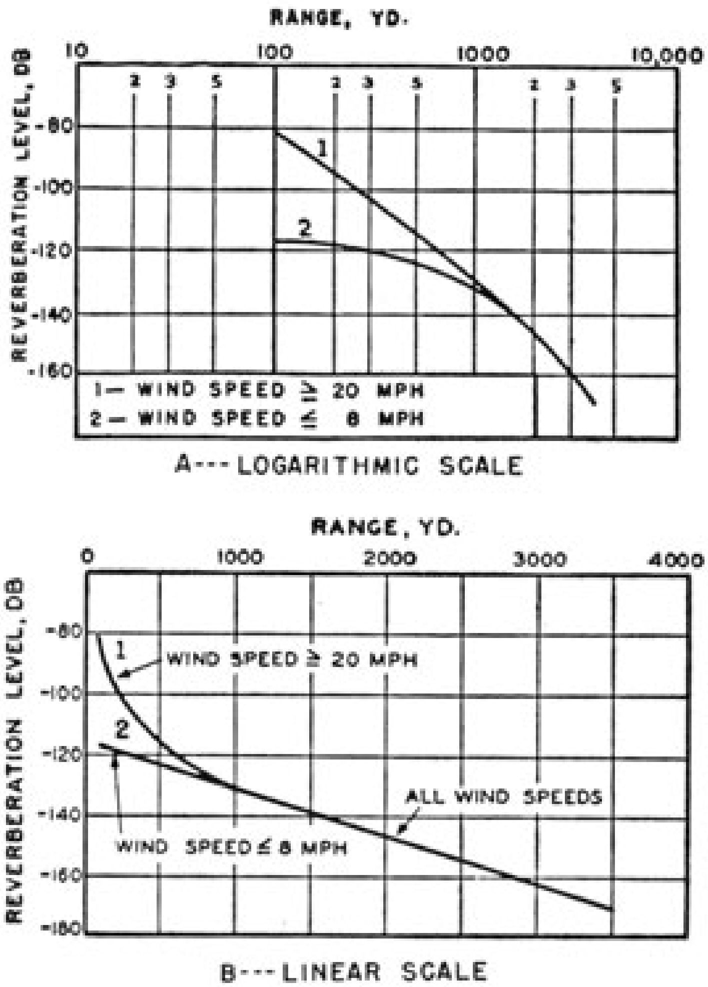


Figure 1-31 -Effect of wind speed on average reverberation level.

more slowly.

The dependence of the short-range reverberation on wind speed clearly indicates that at ranges shorter than 500 yards and at high wind speeds, surface reverberation completely dominates volume reverberation. This conclusion is supported by observations made at nearly the same time with horizontal and vertical beams. At high wind speeds and at short ranges the reverberation levels obtained with a horizontal beam are much higher than those obtained with a tilted beam. Figure 1-32 shows data of this type taken at a wind speed of 17 miles per hour. Points *A* and *B* represent deep scattering layers. Comparison of the two curves shows that in the first 100 yards the horizontal reverberation is about 20 db above the vertical reverberation. Two scattering layers (A and B) are also shown in figure 1-32 at depths of 80 and 400 yards.

At low wind speeds (curve *2* of figure 1-31) the short-range reverberation is volume reverberation. The evidence for this statement is afforded by experiments of the type described in the previous paragraph. When such measurements are made at very low wind speeds, with the sea dead calm, the horizontal reverberation is much lower than in figure 1-32 and agrees well with the vertical reverberation. These measurements indicate that at very low wind speeds, volume reverberation is dominant and surface reverberation is negligible.

Finally, at long ranges (figure 1-31) the reverberation is independent of wind speed. This fact is taken as evidence that at these ranges, volume reverberation always dominates surface

At very short ranges the active surface area is energized by the outer portions of the beam, beyond the angle α; these portions emit sound of a lower intensity than the main beam, and the receiver has a lower response at large angles than on the axis; thus, there is a noticeable drop in the reverberation level. This drop can be calculated from the beam pattern, as shown by the solid curve. At ranges greater than 80 yards the active area is energized by the main beam only, and the measured reverberation levels fit the inverse third-power line closely.

The curves in figure 1-33 must not be regarded as universal. Examples of reverberation curves that show an inverse fifth-power variation of the reverberation with range are frequent. Curve *1* of

reverberation.

Thus, three main conclusions may be drawn regarding deep-water reverberation with a horizontal beam.

1. At short ranges and high wind speeds, surface reverberation is high and dominates volume reverberation (curve *1*, figure 1-31).
2. At short ranges and low wind speeds, surface reverberation is negligible and volume reverberation is dominant (curve *2*, figure 1-31).
3. At long ranges (beyond 1,000 yards), volume reverberation dominates at all wind speeds and is independent of wind speed.

Surface and volume reverberation will now be considered in more detail.

SURFACE REVERBERATION

The discussion of surface reverberation given earlier in this chapter predicts an inverse third-power dependence on range. An example of this dependence is shown in figure 1-33. The data shown in figure 1-33, A, were taken with a transducer, almost nondirectional in the vertical plane and mounted at a depth of 20 feet. Short pings, 6.4 yards long, were used. Wind speed was about 15 miles per hour, so that the resulting reverberation could be identified as surface reverberation. The observed points agree well with the theoretical inverse cube law.

Figure 1-33, B, is a reverberation curve taken the same day under similar conditions. However, the transducer had a pattern in the vertical plane which was highly directional. The axis of the beam was horizontal in both cases. The theoretical curve takes account of the beam pattern.

figure 1-31 is an example. The reason for this rapid decay is not understood.

There is a thermal condition in which the surface reverberation is frequently observed to drop off more rapidly than the inverse cube. This can be explained by conditions of strong downward refraction. The reverberation would be expected to decrease at that range where the sound beam is bent away from the layer of surface scatterers.

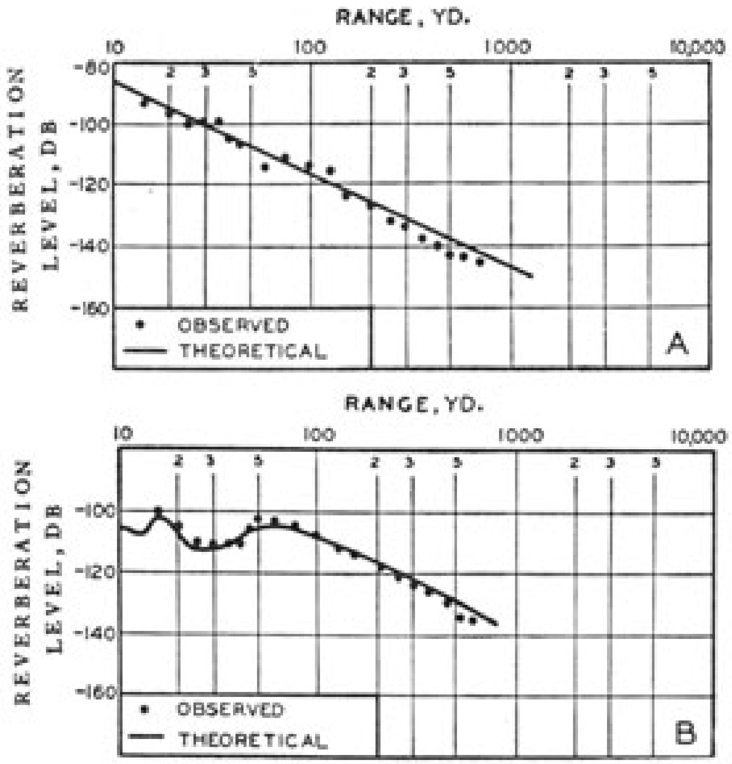


Figure 1-33 -Comparison of observed and calculated surface reverberation. A, Measurements made with a transducer that was almost nondirectional in the vertical plane; B, data taken the same day under similar conditions, but with the transducer turned so that its directivity in the vertical plane was high.

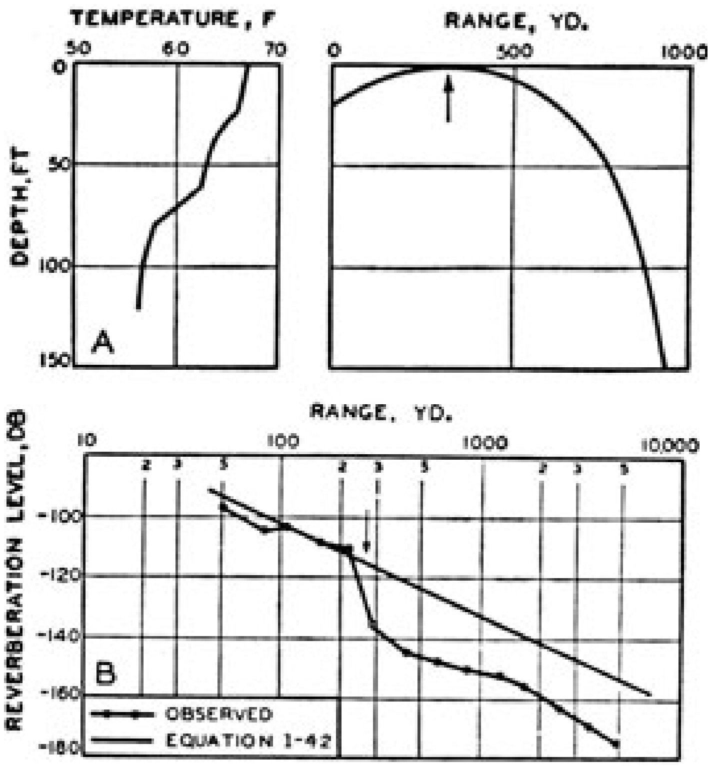


Figure 1-34 -Effect of downward refraction on reverberation. A, Bathythermograph and corresponding ray diagram; B, comparison of observed results with those calculated from simple theory.

Figure 1-34, B, shows an example of this drop, indicated by the arrow. It occurs at about 300 yards; at greater ranges the reverberation level is about 20 db below the value as given by equation (1-42). The data were taken with short (9-yard) pings. The transducer depth was 20 feet and the wind speed 12 miles per hour. The ray diagram based on the bathythermogram in figure 1-34, A, shows that the limiting ray leaves the surface near the 300-yard range (indicated by an arrow), thus affording support for the preceding suggested explanation.

The dependence of surface reverberation on wind speed is very marked at short ranges, as can be seen in figure 1-31. At 100 yards the reverberation level at high wind speeds is some 35 db above that for low speeds, but at 500 yards the difference is only 10 db. The rapid increase of reverberation at 100 yards is seen more clearly in figure 1-35, in which the average reverberation level is plotted against wind speed.

The reverberation level is constant for wind speeds up to 6 miles per hour. This confirms the conclusion that volume reverberation is dominant at these low wind speeds. For wind speeds of 6 to 20 miles per hour the reverberation increases as much as 35 db; the curve then levels off, and above 20 miles per hour there is little further dependence on wind speed. This dependence on wind speed is closely correlated with the roughness of the sea. At 6 miles per hour the wind is strong enough to roughen the surface appreciably; occasionally, wavelets may slough over, but no well-developed whitecaps begin to appear, and when the wind has reached 20 miles per hour the sea is liberally covered with them. When this stage is reached, further increase in whitecaps has no effect on the reverberation.

It has been pointed out that surface reverberation in the ocean rarely exhibits the inverse third-power dependence on range which is predicted by the simple theory. This lack of agreement is found even at short ranges (100 to 500 yards), as shown by the steep slope of curve *1* in figure 1-31. Thus, it is clear that even the average surface reverberation cannot be fitted by equation (1-42) at all ranges. It is possible, however, to apply the equation to the observed reverberation level at one range to obtain the scattering coefficient as a function of wind speed.

VOLUME REVERBERATION

The simple theory of volume reverberation that was

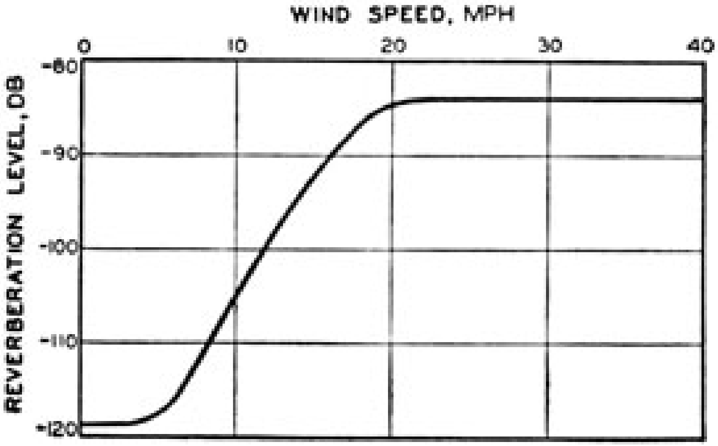


Figure 1-35 -Dependence of reverberation level at short rang (100 yards) on wind speed.

developed earlier and that assumed inverse square transmission and uniform distribution of volume scatterers predicts an inverse square dependence on the range. The first assumption is approximately valid at short ranges but breaks down at long ranges because of refraction and attenuation effects. The second assumption is sometimes valid over a limited region; in general,

---

however, the horizontal stratification of the scatterers invalidates the assumption that they are uniformly distributed. Thus, it is clear that the two basic assumptions made in the simple theory are usually not satisfied.

In order to take account of refraction and the uneven distribution of scatterers, it would be necessary to carry out a volume integration over the active scattering volume at each range. There is insufficient data to warrant such a complex theory. The correction for attenuation, however, is easily made in connection with the calculation of the volume-scattering coefficients. Finally, surface reflection, on the average, raises the reverberation by an additional 3 db. Thus, for a horizontal beam, the theoretical volume reverberation, corrected for surface reflection and attenuation, is

$$RL_v = J_v + 10 \log (mr_o/2) - 20 \log r - 2ar + 3 \quad (1\text{-}45)$$

The importance of attenuation at long ranges is shown strikingly by the large differences between curves *3* and *4*. Thus, at 5,000 yards the attenuation reduces the reverberation level by about 45 db below the inverse square value of curve *3*. Note that the shape of the theoretical curve *4* beyond 1,000 yards is determined largely by the particular value of the attenuation coefficient *a*.

To return to the fit between equation (1-45) and the average curve at long ranges-not only does curve *4* fit the average volume reverberation, but it also fits most individual reverberation curves fairly well. These results indicate that the long-range volume reverberation is due largely to a deep scattering layer. These scattering layers may be colonies of plankton, or fish feeding on it, or bubbles generated by it. Further evidence for this conclusion is afforded by the fact that at short ranges, where the sound beam has not yet reached the deep scattering layer, the observed volume reverberation (curve *2*) falls below the theoretical curve.

It has been remarked that beyond 1,000 yards most individual reverberation curves fit curve *4* closely. This is true over a wide range of oceanographic conditions, with one exception: no significant dependence has been found on wind speed, sea state,

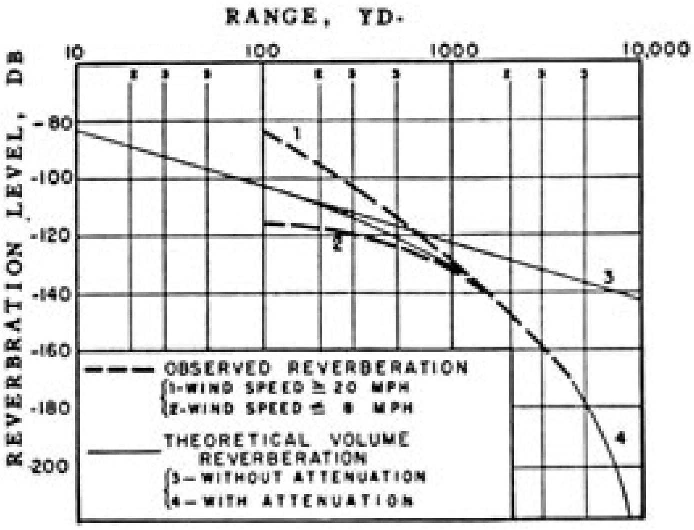


Figure 1-36 -Comparison of calculated and observed volume reverberation, showing the close agreement.

The observed volume reverberation beyond 1,000 yards agrees closely with the theoretical reverberation given by equation (1-45) for typical values of *a* and *m*. This is shown in figure 1-36. Curves *1* and *2* are the observed averages at high and low wind speeds. Curves *3* and *4* were calculated from equation (1-45) using attenuation coefficients of a=0 and a=0.0045 db/yard. The latter is typical of good transmission at 24 kc.

For the remaining parameters the following values were used:

J_v=-25 db

r_o=80 yards

m=10^{-6} yards^{-1}.

location, season, or thermal structure of the ocean.

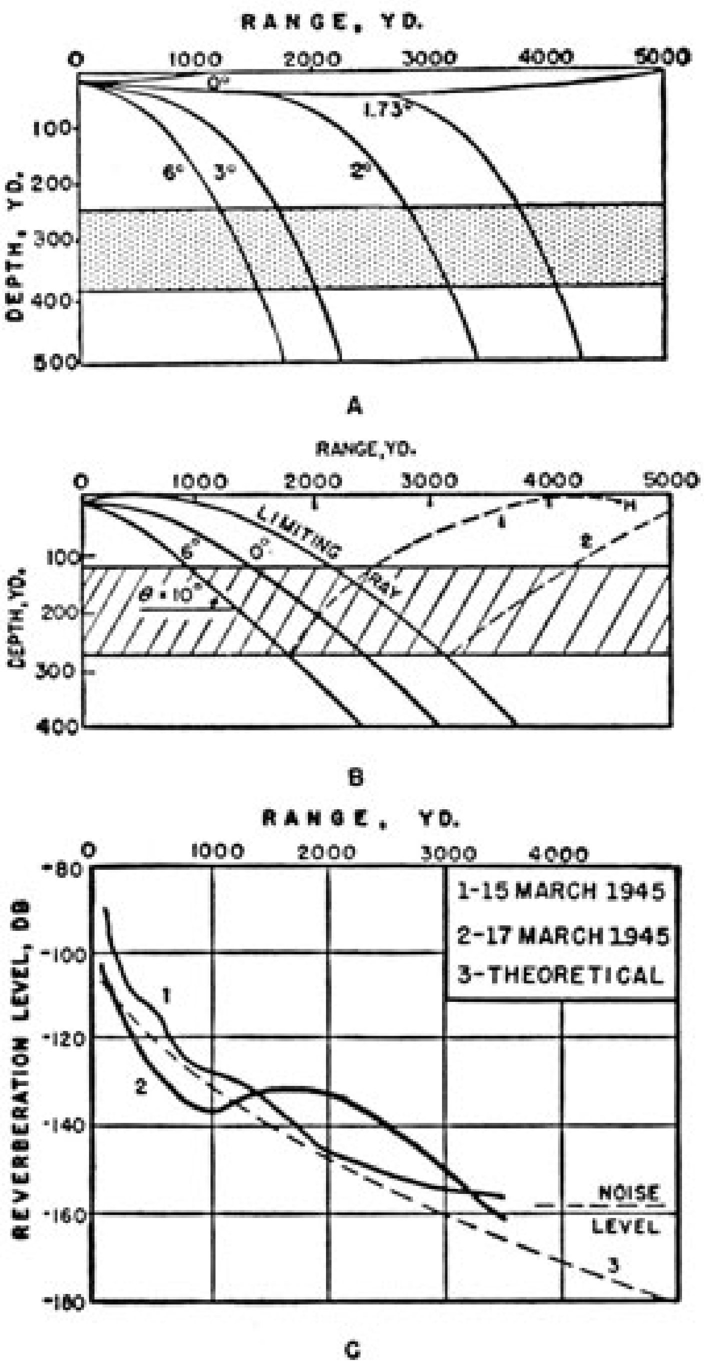
The exception occurs under conditions of extremely sharp downward refraction and provides an interesting check on the importance of the deep scattering layer.

The effect of sharp downward refraction is to concentrate the sound beam into a relatively narrow cone. This produces a maximum in the reverberation curve at the range where the sound beam reaches the layer. An example of this effect is shown in figure 1-37, where refraction and reverberation are compared for 2 days.

On the first day there was a deep mixed layer extending from the surface to a depth of 40 yards. Figure 1-37, A, shows the ray diagram and the deep scattering layer indicated by the shaded portion. The angle shown on each ray is the angle of the ray at the projector, measured downward from the horizontal: the ray of 6° is the effective lower edge of the sound beam. Two days later, on March 17, the same deep layer was still present, but thermal conditions had changed

radically, producing the strong downward refraction shown in figure 1-37, B.

Typical reverberation curves for each day are shown in figure 1-37, C, together with the theoretical reverberation (curve *4*), of figure 1-36. The reverberation observed when there was a mixed layer (curve *1*) agrees well with the theoretical curve *3* between 1,000 yards and 2,500 yards; beyond 2,500 yards it reaches the noise level and flattens out. Curve *2*, on the other hand. observed when there was sharp downward refraction, shows a large maximum near 2,000 yards, corresponding to the range at which the central portion of the sound beam reached the depth of maximum scattering (figure 1-37, B).



At short ranges, curve *1* rises steeply with decreasing range. This rise is due to surface reverberation and is to be expected, because the data were taken at a wind speed of 20 miles per hour. The data of curve *2* were taken at a wind speed of 12 miles per hour; it shows a corresponding rise in the reverberation level at very short ranges (100 yards), but there is a minimum when the sound beam has left the surface and has not yet reached the scattering layer. When the lower edge of the beam reaches the deep layer (about 1,000 yards) the reverberation begins to increase with increasing range, culminating in the main maximum.

BOTTOM REVERBERATION

Because in echo ranging, the transducer is generally near the surface, the sound scattered back from the bottom provides an important contribution to the reverberation only in shallow water; here, however, it may well be the dominant factor in limiting the range from which detectable echoes can be obtained.

When the reverberation is caused by the surface, it is the state of the sea that determines the intensity of the scattered sound; bottom reverberation levels may be expected to depend on the character of the sea bed. In practical work four types of bottoms are recognized-(1) *rock*, (2) *sand*, (3) *mud and sand*, and (4) *mud*. The criterion of classification is the size of the particles constituting the sea bed, as determined by examining samples obtained by sounding with special devices. Recently, also, techniques of underwater photography have been perfected and have proved useful in studying the bottom. The difference in reverberation intensities among these types of bottoms are discussed as follows in connection with the discussion of bottom-

Figure 1-37 -Comparison of reverberation and refraction for 2 days. A, Ray diagram for March 15, 1945; B, ray diagram for March 17,1945; C, observed reverberation for these 2 days compared with calculated values.

scattering coefficients.

It is obvious, from the discussion of figure 1-26 that any bottom reverberation will be combined with volume reverberation and, when a horizontally directed beam is used, with surface reverberation. Thus, from the geometry of the experiment, it should not be expected that the measured levels of predominantly bottom reverberation have the levels predicted by the simple theory expressed

28

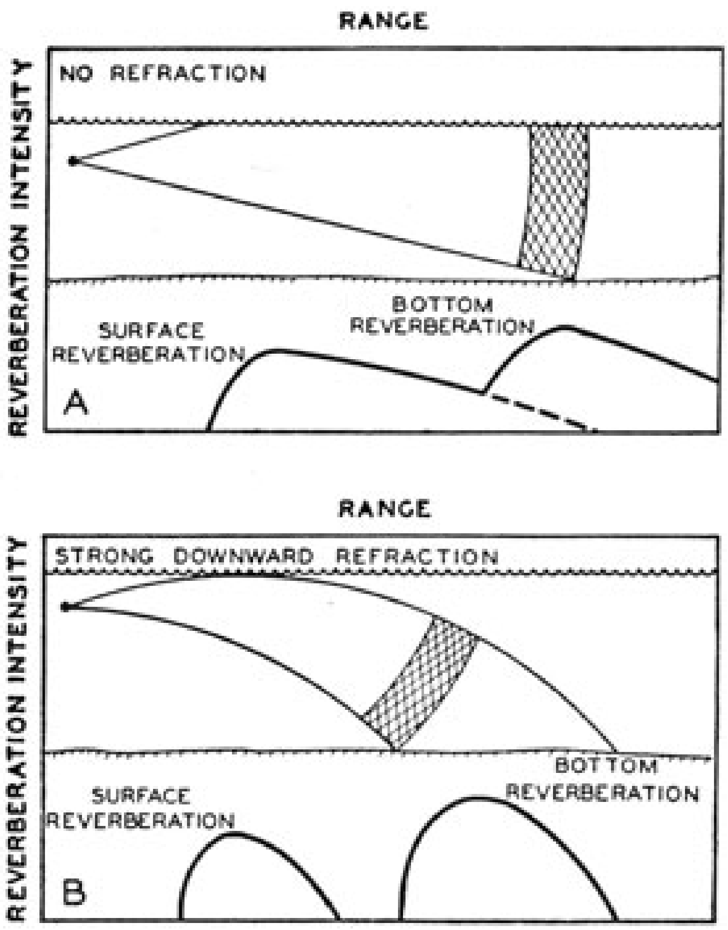


Figure 1-38 -Schematic diagrams illustrating surface and bottom reverberation. A, No refraction; and B, strong downward refraction.

by equation (1-44). It will be recalled that equation (1-44) applies to bottom reverberation as well as to surface reverberation. In shallow water over a bottom that scatters strongly, such as rock, the bottom reverberation may be so much greater than either the surface or volume reverberation

surface increases the intensity of the reverberation fourfold or, expressed in decibels, raises the reverberation level 6 db.

The effects of the refraction of the sound are more difficult to evaluate. The bending and distortion of the sound beam affects the intensity of the bottom reverberation in several ways. If the beam is bent sharply downward, the sound strikes the bottom at a shorter range and may be more concentrated; the surface reverberation decays very rapidly, And the bottom reverberation may be more intense. This is illustrated schematically in figure 1-38, A and B.

that a rough check of the theory is possible. If this is attempted, however, the simple inverse square loss will not provide a very reliable guide to the total transmission loss. Consideration must also be given to the loss due to attenuation (absorption and scattering).

In addition, the effect of the surface in reflecting the sound incident on it toward the bottom, from which it may be scattered back to the transducer either directly or by way of the surface a second time, must be taken into account. If the surface is considered to act as a perfect reflector, direct sound at the bottom evidently will be doubled, and thus the intensity of the scattered sound will be doubled. Moreover, the reflection of this scattered sound from the surface causes the intensity at the transducer to be doubled. Hence, the

239276°-53-3

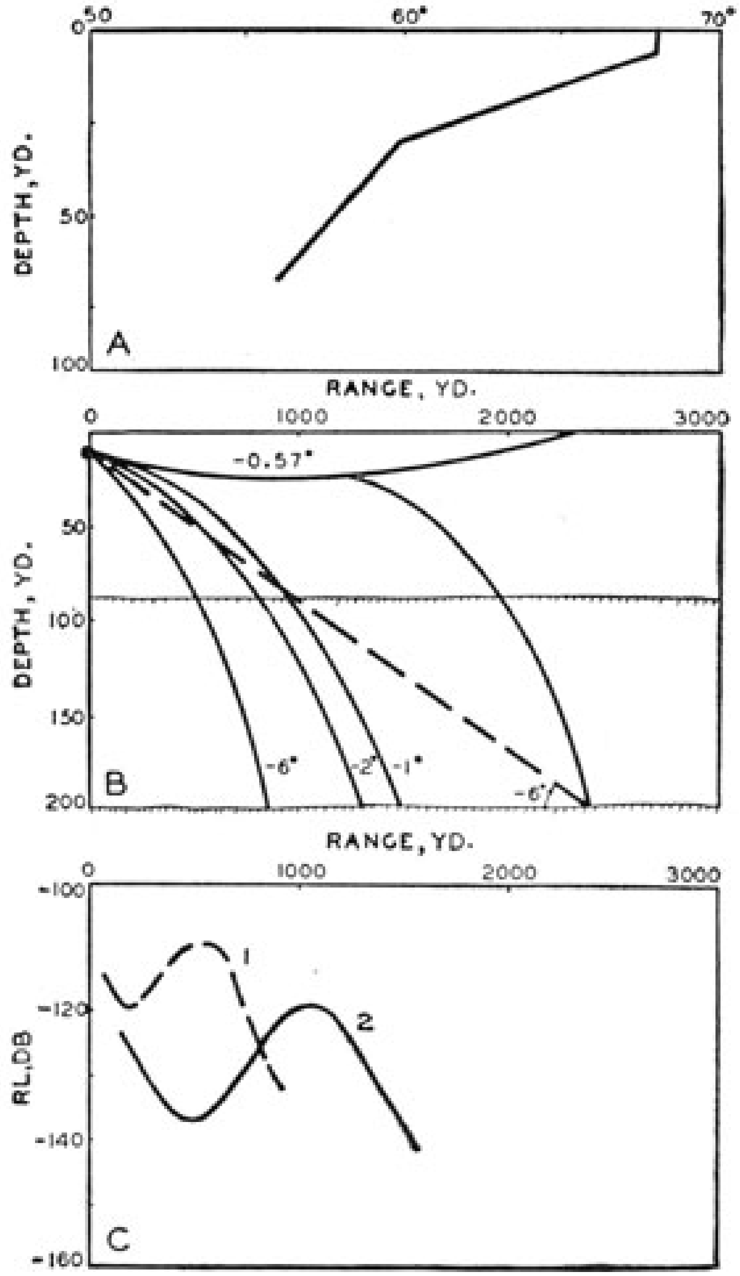


Figure 1-39 -Data of an experiment illustrating the conditions shown in figure 1-38. A, Bathythermogram; B, ray diagram; C, observed reverberation at water depths of 87 yards (curve *1*) and 210 yards (curve *2*).

It is clear from figure 1-38, A, that, if the beam is not refracted downward, surface reverberation will be received continuously after the beam first strikes the surface. On the other hand, as seen from figure 1-38, B, a sharply refracted beam strikes the surface in a limited area only; the surface reverberation consists of a burst of sound that dies away very rapidly, to be followed by a second burst of sound as the bottom reverberation comes in.

These effects are shown in figure 1-39. The bathythermogram shows the thermal pattern of the sea when the reverberation shown by curves *1* and *2* in figure 1-39, C was measured. The two curves represent reverberation in two different depths of water (87 and 210) yards . The bending

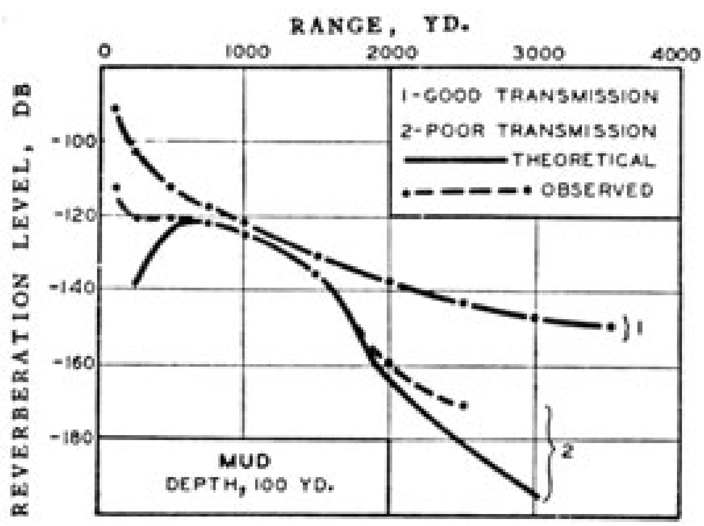


Figure 1-40 -Comparison of calculated and observed reverberation in shallow water over mud bottom.

and distortion of the beam is seen in the ray diagram; a dotted line is drawn showing the path of the ray to be -6° in the absence of refraction. The refracted ray of -6° strikes the bottom at ranges shorter by 500 and 1,200 yards at the two depths. Moreover, the beam is concentrated between the ray of -1° and the upper limiting ray. The expected rapid decrease in reverberation as the upper half of the beam strikes the bottom is shown in curves *1* and *2* of figure 1-39, C. They

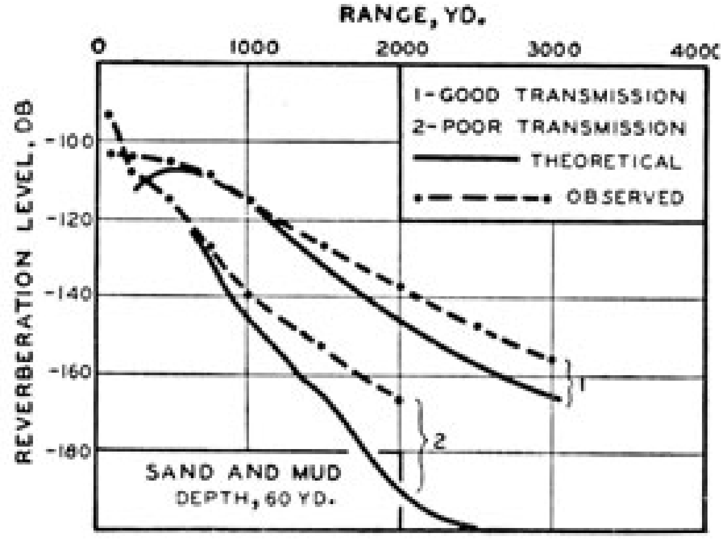


Figure 1-41 -Comparison of calculated and observed reverberation in shallow water over sand and mud bottom.

Of the terms in this equation, J_s, and r_o, are known; the value of H_s, must either be predicted or else obtained by making transmission measurements as nearly simultaneously with the reverberation runs as practicable. The magnitude of the scattering coefficient *n* is, of course, not known; hence that value of *n* which best fits the experimental points is considered to be the appropriate scattering coefficient. In determining the best fit give the greatest weight to the region of the graph between 500 and 1,000 yards.

The agreement between the calculated and observed reverberation is illustrated by figures 1-40 through 1-43.

Each of these figures exhibits results of measurements taken over a particular bottom type. Two theoretical curves are drawn for each bottom type except mud, in which transmission data were available only for poor transmission. The values

also show the expected rapid decrease of the surface reverberation. The difference in levels between the two curves is due to the difference in ranges to the bottom in the two cases.

A large number of bottom-reverberation records have been plotted and compared with the graph of *RL*, as a function of *r*, as given by equation (1-44).

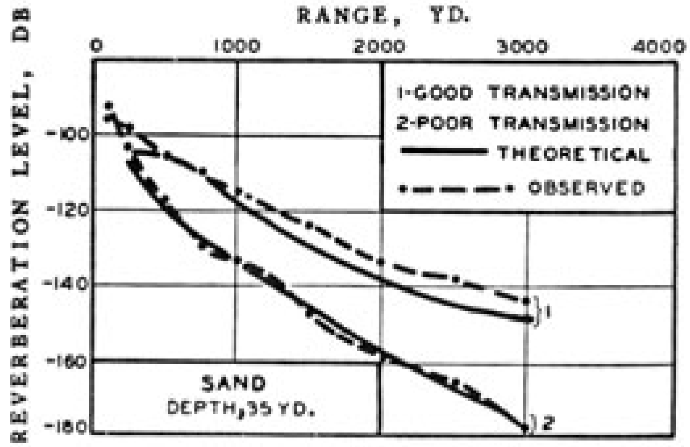


Figure 1-42 -Comparison of calculated and observed reverberation in shallow water over sand bottom.

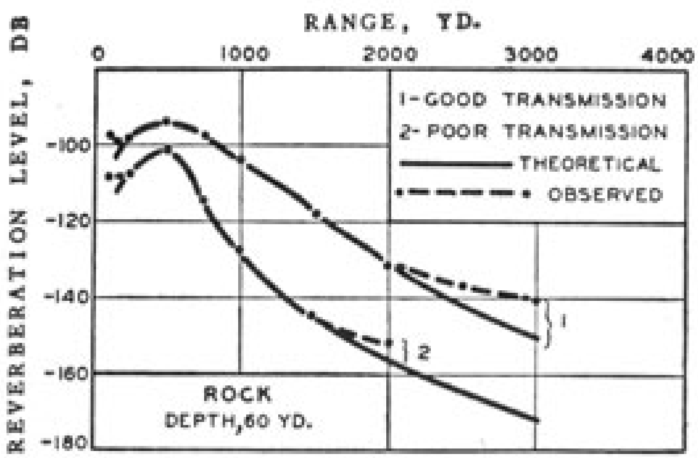


Figure 1-43 -Comparison of calculated and observed reverberation in shallow water over rock bottom.

of *H*, were measured in conjunction with the reverberation measurements.

The experimental points represent averages of from 5 to 30 reverberation runs. The quartile deviation is about ± 5 db on the average. All measurements were made using 80-yard pings.

It should be stressed that these experimental data are not to be considered as representing bottom reverberation generally in shallow water. The measurements that yielded them were taken over

TABLE 1 -*Bottom Scattering Coefficients*

Bottom type	10 log *n*
Rock	-22
Mud and mud sand	-30
Sand	-34

at longer ranges is not known. See table 1 for coefficients of bottom scattering.

FORWARD SCATTERING

The simple refraction theory predicts a "black" shadow zone under conditions of strong downward refraction. This is not confirmed by transmission runs of 24 kc made in deep water off the southern California coast. Sound of very low intensity is observed out to ranges of 5,000 yards (some 4,000 yards beyond the limit of the strong direct sound field), which can be explained neither as direct nor as bottom reflected sound. Instead there appears good reason to believe that it is sound scattered in the forward direction from a deep scattering layer.

The general good agreement between the theory and

particular small patches of particular bottom types. The curves serve, however, to convey a fairly realistic picture of the main features of reverberation in shallow water.

The four figures have certain features in common. The figures are good for all ranges less than 1,500 yards. Beyond this range the measured reverberation is consistently higher than the calculated one. Over mud bottom, the difference is 10 db at 2,500 yards over sand and mud; it is 20 db at 2,000 yards. The reason for this divergence the observations, in the few cases that can be checked, indicates two main conclusions:

1. Sound observed in the shadow zone under conditions of sharp downward refraction is forward scattering from the deep scattering layer.
2. The scatterers in the deep scattering layer are approximately isotropic; that is, they scatter sound nearly equally in all directions.

Sonar Home Page

Next Chapter

CHAPTER 2

TRANSMISSION OF SOUND IN SEA WATER

Introduction

A study of those factors that affect the propagation of sound energy in the sea and determine the subsurface path of a sound beam after it is transmitted from a transducer is essential to a full understanding of the sonar problem. Some of the factors that make up the transmission anomaly were discussed in chapter 1. The influence of temperature conditions in the upper layers was especially noted.

The variability in the transmission loss was first observed in actual echo-ranging operations at sea. In certain areas, the ranges achieved in the afternoons of clear, relatively calm days were found to be less than those obtained in the mornings. Temperature gradients change from season to season, from day to day, and even from hour to hour and place to place. Numerous explanations were advanced, but the true reason was discovered by the Woods Hole Oceanographic Institution as the result of experiments performed in cooperation with the Navy.

In many respects the ocean is very similar to the atmosphere, and thus there is a close analogy between oceanography, on the one hand, and meteorology and climatology on the other. One may speak of the subsurface weather and of its seasonal and diurnal changes, of the subsurface climate, and the annual and seasonal averages of the components of the weather.

The analogy between oceanography and meteorology holds true further in that one of the practical objectives of the oceanography of underwater sound is the forecasting of subsurface weather. The study of subsurface weather was neglected until its importance for underwater acoustics was recognized.

General Processes and Their Interaction

The outstanding characteristic of weather, both in the air and under the sea, is its changeability. This changeability is the outcome of a complex set of processes, which are continuously in action. Sometimes one of these processes may dominate all others; more often, several exert appreciable influences on the resultant.

There are at least 10 such processes that cause the temperature gradients in the upper layers of the ocean to change. They can be grouped conveniently into four general processes-(1) heating, (2) cooling, (3) mixing, and (4) flowing at gradients that are revealed by bathythermograms. Each process is caused by a variety of factors. All four, however, are affected by the condition of the atmosphere at the ocean surface. The immediate effect of each process is to alter the dynamic state of the surface layers.

Table 2 presents an outline of the general processes, with their causes and dynamic effects. A characteristic complication is illustrated by processes 2 and 3-cooling makes the surface layer unstable, and instability in turn causes mixing. In the same way, strong currents may cause turbulence,

a speed different from that of the underlying water. All four processes are closely interrelated but each has its own characteristic effect on the temperature which again results in mixing. There are many other chains of cause and effect linking all the processes.

TABLE 2 -*Outline of Processes Influencing Temperature*

General process	Cause	Dynamic effect
Heating	Sunshine and/or warm, moist air.	Stability of surface layer.
Cooling	Evaporation, back radiation, and/or cold, dry wind.	Instability of surface layer.
Mixing	Wind and waves, instability and/or turbulence.	Neutral stability of surface layer.
Flowing	Wind and waves, internal waves, and/or currents.	Variable; turbulence if strong.

STRATIFICATION OF THE OCEAN

Bathythermograms show that the ocean is more or less stratified. Two points separated by several hundred yards but at the same depth beneath the surface have practically the same temperature. If the ocean were in equilibrium, this stratification would be complete. The warm, lighter water would be at the surface; the cooler, heavier water would be at the bottom; and the boundaries between strata would be horizontal surfaces. Such an equilibrium is disturbed by three of the four general processes. The observed stratification is thus the result of other processes tending to bring the ocean to equilibrium.

DENSITY

differences may sometimes dominate the density distribution so that a layer of cold dilute water may overlie warmer water of high salinity.

THERMAL STRUCTURE AND STABILITY

The concept of stability is a convenient one to apply. Stability depends on the rate at which density increases with depth. If the temperature in a layer decreases rapidly with depth, as in the thermocline, the layer has high stability, for the density increases rapidly. On the other hand, a layer in which the density decreases with depth is unstable and exists only transiently.

Mixing processes are retarded by high stability; thus wind of a given strength may easily mix a surface layer in which the temperature gradient is small and the stability is low. The same wind may have little mixing effect if the temperature gradients near the surface are large. The development of a sharp thermocline tends to retard mixing to greater depths.

A completely mixed isothermal layer has neutral stability. Cooling at the surface increases the density of the surface layer, Evaporation, because of the cooling and the increase in salinity that accompany it, also increases the density of the surface layer. Hence these processes tend to make the density of the surface water greater than that of water immediately below it and to produce a condition of instability. This unstable density distribution near the surface results in convective mixing.

The stability can be estimated from a bathythermogram if the salinity gradients are

It is a general hydrodynamic principle that when a mass of fluid is in stable equilibrium under the force of gravity its density must everywhere increase in the downward direction and be constant in every horizontal plane. A commonplace illustration of this principle is furnished by a bottle containing oil and water.

The density of sea water is determined primarily by its temperature and salinity. The changes due to temperature are the largest, just as with the velocity of sound. However, salinity has a proportionately greater effect on the density of sea water than on the velocity of sound.

In the open ocean, where the salinity is practically constant, the lighter water almost always is the warmer water, and it is to be expected that the temperature either remains constant or decreases with increasing depth. Near the shore, salinity assumed to be negligible. Density decreases with increasing temperature and for most practical purposes the isotherms on the bathythermogram grid can be interpreted as lines of equal density. The slope of the temperature trace is therefore a measure of the rate of change of density with depth-that is, of stability. If this fact is kept in mind, the bathythermograph traces can be interpreted in terms of the four major processes.

LABORATORY EXPERIMENTS ON STRATIFICATION

Negative thermal gradients are very stable because there is little exchange of heat between neighboring layers unless they are mixed by some General Processes and Their Interaction

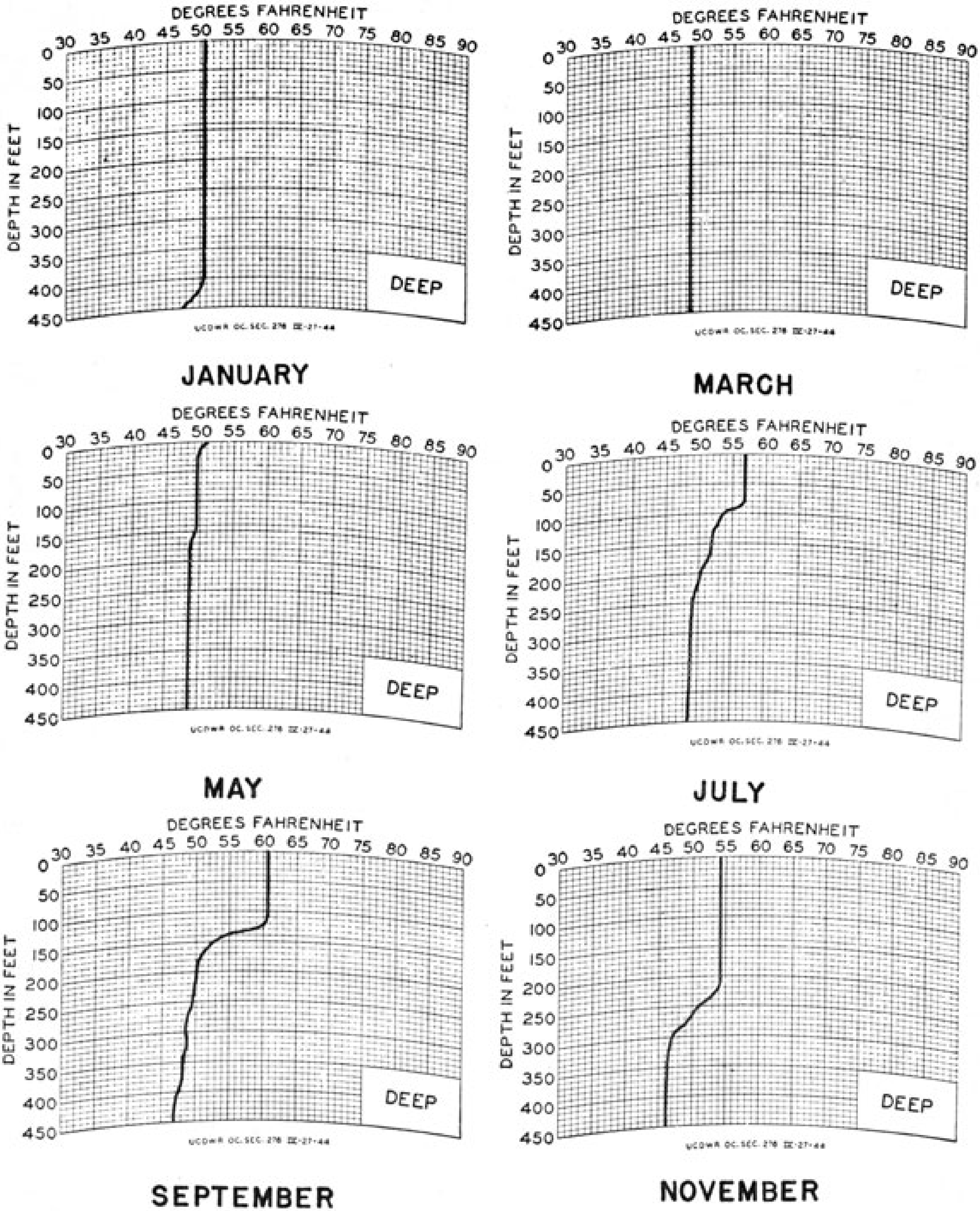


Figure 2-1 -Annual cycle of ocean temperature gradients.

stirring action. This fact is shown readily by laboratory experiments. If a tank is partly filled with warm water, and if water of room temperature is then run in through a hose lying on the bottom, the warm water floats on the colder. Thermometers placed in the two layers show that the cooler water is not heated by the overlying warm water.

This stability of layers when the temperature gradients are negative is in marked contrast to the instability of positive temperature gradients. In the experiment of warm and cold layers of water in a tank, the surface of the warm layer may be cooled by blowing a gentle stream of cold air over it. The cooling of the layer at the immediate surface causes it to become heavier than the water beneath it. Consequently it sinks and in so doing mixes with and cools the underlying water. Two thermometers at different depths in the warm layer show that cooling proceeds nearly simultaneously at all depths, without the development of large positive temperature gradients. The mixing that accompanies cooling is called *convective overturn*.

EFFECTS OF THE GENERAL PROCESSES ON TEMPERATURE STRUCTURE

Heating

The progressive or intermittent effects of the four processes-heating, cooling, mixing, and flowing-lead to the complicated and variable conditions illustrated in figure 2-1. The manner in which any one of these processes operates individually to change the bathythermogram is shown in figure 2-2. The change in temperature distribution produced by solar heating is illustrated by curves 1, 2, and 3 in figure 2-2, A. Initial conditions,

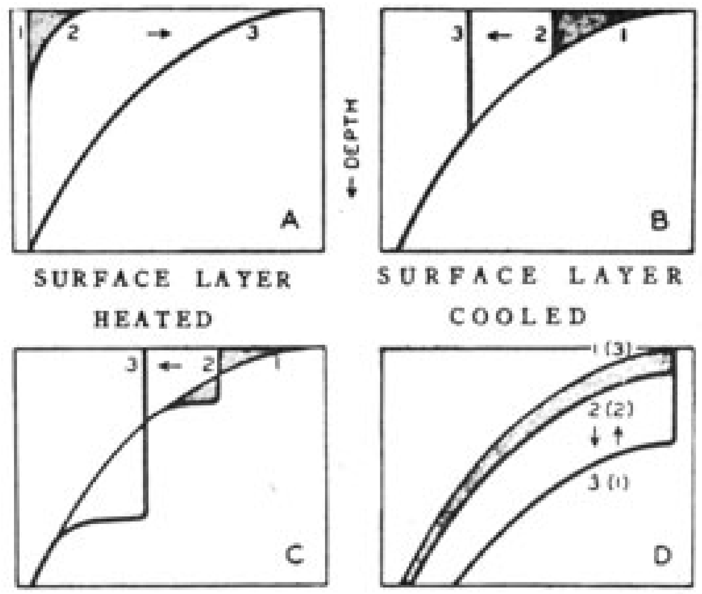


Figure 2-2 -Manner in which the general processes working individually change the bathythermogram. A, development of negative gradient by heating of surface layer; B, development of isothermal surface layer by cooling; C, development of isothermal surface layer by mixing; D, effect of current.

which is assumed to be the same as curve 3 in the preceding diagram, surface cooling with its accompanying convective overturn produces curve 2 and ultimately curve 3. If continued long enough, it would finally produce completely isothermal water. Although the cooling takes place at the surface, measurable positive gradients do not develop because of the mixing involved in the convective overturn. Winds hasten this mixing process, but convective overturn takes place even in very calm weather. Theoretically, the upward transfer of heat must be associated with slight positive gradients, but such gradients are so small that they usually escape detection.

Mixing

The result of vigorous mixing by the wind, when there is no gain or loss of heat by the surface layer,

indicated by curve 1, are assumed to be isothermal. The absorption of heat, accompanied by some mixing, results in curve 2 and finally curve 3. Negative gradients extending from the surface downward are characteristic of recent heating. The negative gradients, and consequently the stability, become greater as the amount of mixing that occurs during the heating becomes smaller. Under these conditions, wind is the principal cause of mixing.

Cooling

The cooling that takes place during the night and during the winter is essentially a reversal of the heating process. In curve 1 (figure 2-2, B), is illustrated in figure 2-2, C. Note that in this example surface isothermal layers develop just as they did in figure 2-2, B, and the surface temperature decreases, but the temperature distribution immediately below the mixed layer is different. The wind mixes warm water with cooler water beneath it, increasing the temperature at intermediate depths, and thus produces a very sharp thermocline instead of retaining the initial gradients present when cooling is the primary cause of the mixing. Curve 1 in figure 2-2, C, is the same

as curve 1 in figure 2-2, B, but the result of wind mixing without cooling produces distributions quite different from those resulting from cooling alone. Obviously, conditions intermediate between those of figures 2-2, B, and 2-2, C, often develop, because cooling and wind mixing can occur simultaneously.

Flowing

The effect of addition or removal of water by currents is illustrated in figure 2-2, D. Curves 1, 2, and 3 show the development of an isothermal layer; curves (1), (2), and (3), of a negative gradient. The transfer of water can be produced by various causes, such as winds. If warm surface water is carried over the top of cooler water, a progressive change in temperature distribution may occur, as illustrated by curves 1, 2, and 3. If warm surface water is removed, the reverse sequence, indicated by curves (1), (2), and (3), may develop. Note that the gradients remain unchanged and are merely lowered or raised. Internal waves,. which periodically raise and depths greater than from 600 to 700 feet. Below these depths, stable stratification exists at all times, and the only changes are due to slow seasonal currents. This deep region is therefore characterized by the so-called *permanent thermocline* or negative temperature gradient.

The density of sea water increases with decreasing temperature down to the freezing point (about 28.5° F), which sets a lower limit for the temperature in the sea. Below 6,000 feet the temperatures everywhere are less than about 37.5° F and decrease with depth. They also decrease toward the south, where the coldest water is formed.[1] The circulation of the deep, cold water is exceedingly slow, probably of the order of 1 foot per minute. For all practical purposes the conditions in the deep sea do not change with time; they do, however, vary slightly from one region to another. In any one locality below about 3,000 feet the temperature decreases slowly and the salinity is either constant or increases slightly with depth.

Annual Cycle

lower the thermocline, can cause similar effects in a very short time. These waves may be single or have a well-defined periodic character and are accompanied by single or periodic surges of current.

TEMPERATURE DISTRIBUTION

The four general processes all involve passage of time-that is, *continued* heating, cooling, wind mixing, or flowing produces *progressive* changes in the temperature distribution. In the sea the temperature distribution in a given locality is the result of interplay of all four processes. For a limited time, such as during one afternoon, one of them may dominate, so that the temperature conditions near the surface are the result of heating, cooling, wind mixing, or currents. The complicated distributions illustrated in figure 2-1 are usually the result of intermittent action of the four general processes.

Thermal Structure at Great Depths

All these processes except the flowing originate at the sea surface, and their effects are propagated to greater depths by convective overturn or mixing or both. These effects are rarely noticeable at

In middle and higher latitudes there is a marked annual cycle in temperature conditions. The cycle can be observed in figure 2-1, which is based on bathythermograms taken in the open ocean, in latitude 40° N in the North Pacific.

It is convenient first to consider conditions in March. The isothermal layer then is more than 450 feet thick, and is produced by cooling and by mixing induced by winter storms. In May some heating of the surface layers occurs, and mixing by winds produces an upper isothermal layer of a slightly higher temperature than the original; thus, there is a small thermocline at a depth of about 150 feet. The negative gradient at the surface probably represents heating during the day and is either obliterated by wind mixing or disappears during the night because of cooling and convective overturn.

Progressive heating continues through the summer months so that the temperature near the surface increases, as shown by the July and September bathythermograms; but wind maintains a mixed layer with a rather sharp thermocline, which increases in depth as the season progresses.

---

[1] H. U. Sverdrup, M. W. Johnson, and Richard H. Fleming, *The Oceans*, New York, Prentice-Hall, 1942.

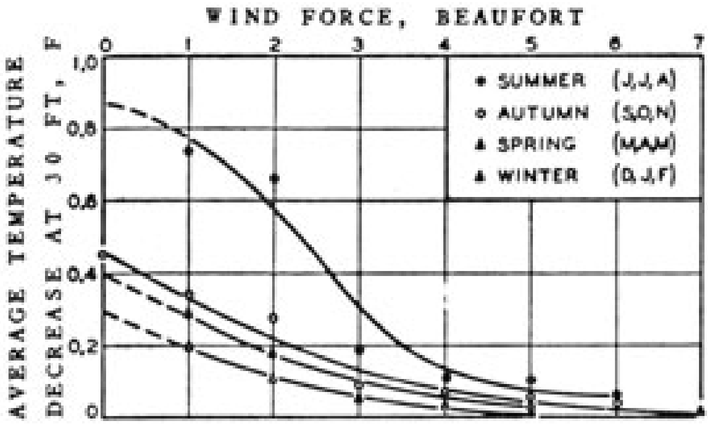


Figure 2-3. - Effect of wind on the average temperature gradient in the surface layer during various seasons.

In the fall, cooling once more exceeds heating; the surface isothermal layer becomes cooler; and, with the added effect of strong winds, the thermocline goes deeper until in January it is below 400 feet. Cooling and mixing continue until about March.

In general the systematic seasonal changes are subject to modification by local weather conditions. The mixing of the surface layer by wind is especially important in this connection. In figure 2-3 the average temperature decrease in the top 30 feet is plotted for each season as a function of wind force. High winds can practically obliterate the seasonal trend.

Diurnal Cycle

The diurnal cycle in temperature conditions is in many ways a miniature replica of the annual cycle, but it must be remembered that progressive heating occurs during the spring and summer and that progressive cooling and mixing occur during the fall and winter. Consequently, the daily cycle sometimes is practically obliterated by the progress of the seasonal changes.

Four selected examples of diurnal changes are given in figure 2-4. The data are from the open action of the wind caused a mixed layer to persist near the surface throughout the day. The layer was so shallow, however, that poor sonar conditions prevailed during the afternoon. During the night, cooling and mixing resulted in isothermal conditions to a depth of 50 feet.

The series shown in figure 2-4, B, is an example of heating on a day with light winds when negative gradients extended to the surface during the late morning and afternoon. Beginning at 1800, a mixed layer was present and cooling continued during the night. An observation at 0600 the next morning showed a small positive gradient which had disappeared by 0800.

The series shown in figure 2-4, C, covers a period of approximately 48 hours with variable winds. No progressive heating is noticeable, and there is a return to isothermal conditions each night.

ocean and are based on bathythermograms taken over periods of from 23 to 48 hours during the summer. Each set has been adjusted so that the temperature at a depth of 50 feet is used as the reference. The heating is indicated by shading.

The series shown in figure 2-4, A, was taken during a day when winds averaged force 3. Although heat was added to the water, the stirring

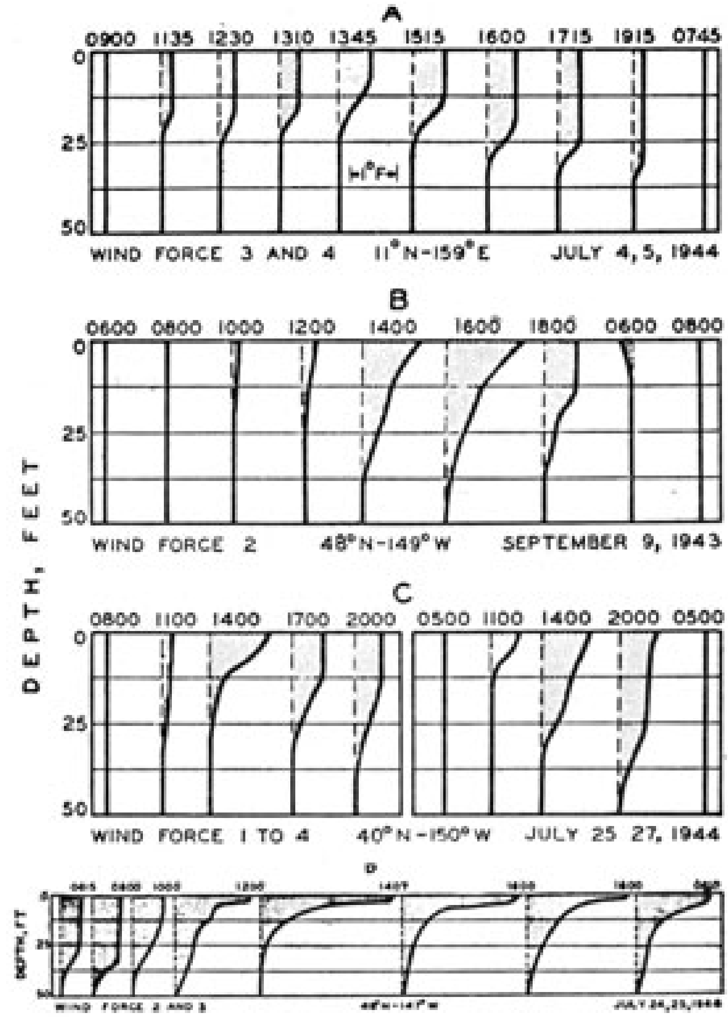


Figure 2-4 -Diurnal cycle of ocean temperature gradients. A, Persistent mixed surface layer; B, typical diurnal cycle with light winds; C, variable winds with changeable pattern; D, persistent negative gradients.

The series shown in figure 2-4, D, is an example of heating when a negative gradient existed early in the morning. The shallow isothermal surface layer had practically disappeared at noon; the gradient became progressively more pronounced during the day and persisted during the following night.

As in the annual cycle, high winds can obliterate the daily cycle in the upper 30 feet. This fact is shown in figure 2-5.

Afternoon Effect

In general, strong negative surface gradients are most common in the afternoon and produce what is called the *afternoon effect*. The gradients reach a maximum about 1600 and a minimum about 0600. Because solar radiation is greatest in the summer, such gradients are more common during the summer than during the winter.

This simple explanation is essentially correct but fails to provide an explanation of the geographical

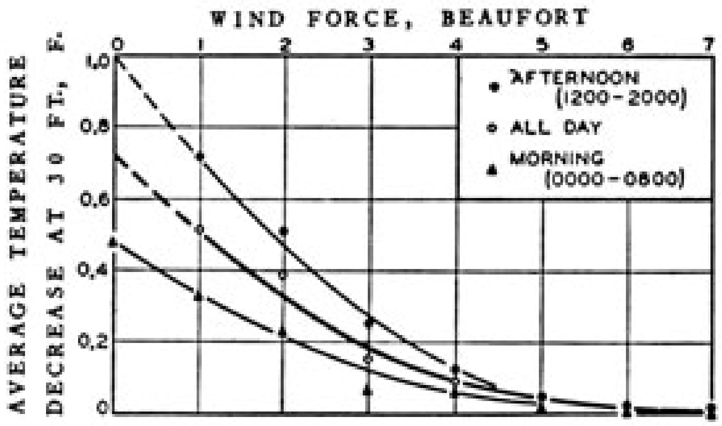


Figure 2-5 -Effect of wind on the average temperature gradient in a surface layer at various times of day.

distribution effect. Instead of being most frequent at the equator, where solar radiation is greatest, the afternoon effect is actually less frequent there than in high latitudes. Solar radiation is undoubtedly the primary cause of the negative surface gradients, but the magnitude of its effect is modified by the other three factors, especially wind mixing and evaporation.

Although in the open ocean, afternoon effect is most frequent in high latitudes, this principle does not necessarily apply inshore. The waters off southern California, for example, are noted for the prevalence of afternoon effect.

Analysis of the Four Processes

The preceding paragraphs have indicated the general types of temperature distribution encountered in the sea and the four major processes that affect the temperature conditions. The causes of temperature conditions will now be discussed.

HEATING AND COOLING

The temperature structure of the ocean is determined primarily by its heat content, which is a constantly varying quantity. There is a continuous exchange of heat at the surface of the ocean. The ocean *receives* heat by *absorption* of the sun's radiation and by the *condensation* of water vapor in the air when the water is colder than the air. The ocean *loses* heat by *radiation* to the sky, by *evaporation* of water vapor when the water is warmer than the air, and possibly by *conduction*. Of the received heat, by far the largest quantity is due to incoming solar radiation. Over the ocean as a whole incoming solar radiation is balanced by the cooling resulting from reradiation and evaporation. The effects of other processes are comparatively negligible.

Incoming Radiation

The incoming radiation includes the invisible infrared and ultraviolet as well as the visible light. Because it is received from the sun and the earth's atmosphere it obviously varies with latitude, time of the year, time of day, and atmospheric conditions-particularly the cloud cover. The total energy received during the year decreases with increasing latitude. In the lower latitudes of the tropical regions the seasonal variation is small, but with increasing latitude the difference between the amounts received during the summer and winter becomes very great. The effect of clouds is very pronounced-a heavy cover of cloud may reduce the incoming radiation to less than 25 percent of that received on a clear day.

Direct heating of the water by the sun is limited to relatively shallow depths (fig. 2-6). Only about 3 percent of the radiation penetrates below 300 feet and over 50 percent (all of the infrared) is absorbed in the first few inches. If there were no compensating heat losses and no mixing, fantastically high surface temperatures and extremely sharp negative gradients just below the

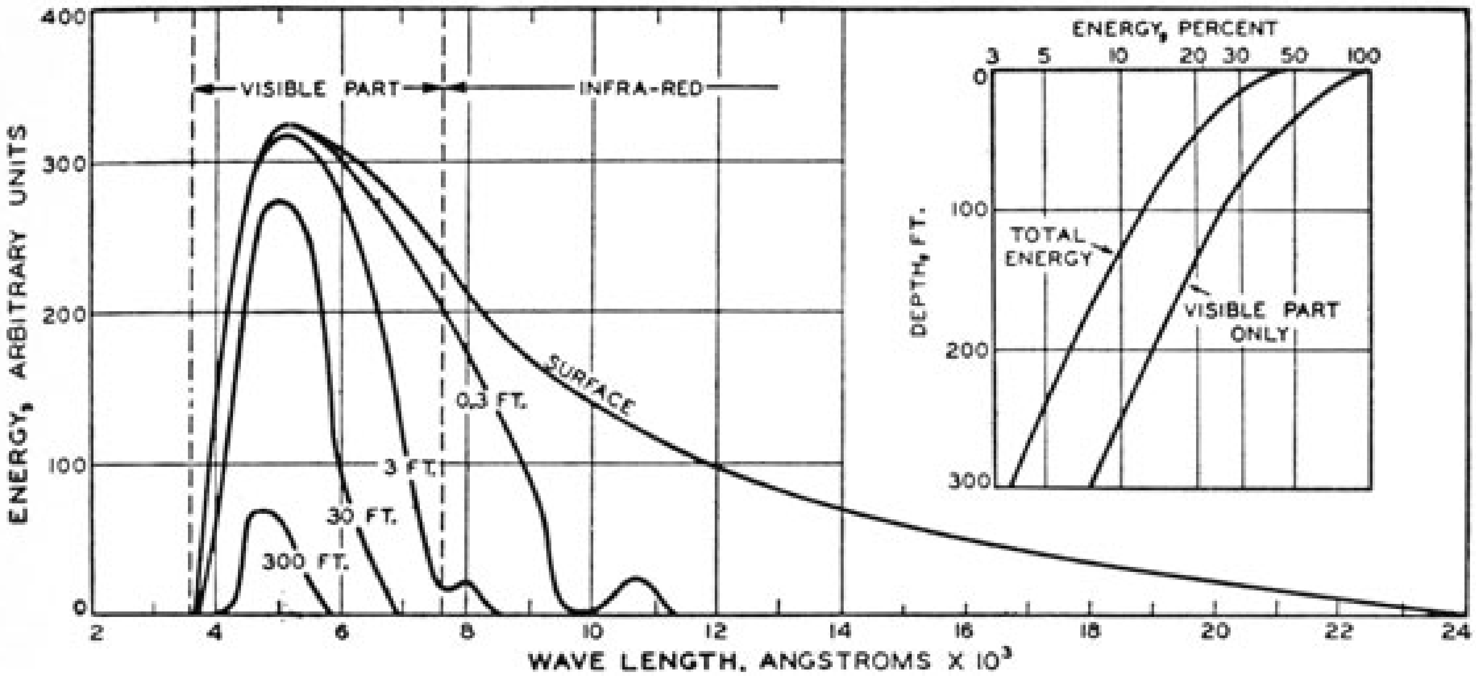


Figure 2-6. -Spectrum of radiant energy at various depths in the ocean. Insert: Percentage of incident radiation reaching various depths.

surface would occur. The penetration of light varies somewhat from place to place depending on the amount of suspended debris and organic pigments in the water. The foregoing discussion applies to the open ocean. Near shore and in areas of vigorous plant growth the water is practically opaque to all wavelengths.

Besides the direct solar radiation, the sea surface also receives some infrared from the air. Although the air is an appreciable source of heat, it is customary to subtract it from the corresponding infrared radiation emitted by the sea surface.

Effective Back Radiation

The excess of infrared emitted by the sea surface over that received from the air is called *effective back radiation*.[2] Effective back radiation balances somewhat less than one-half of the incoming solar radiation, on the average. It decreases with increasing water temperature and with increasing losses from back radiation occur in the uppermost fraction of an inch of the water and are transmitted to greater depths by convective overturn and wind mixing.

Evaporation

Evaporation depends primarily on the temperatures of the water and the air, on the humidity, and on the wind strength. Evaporation can be understood best by considering the process as one of transfer of water vapor away from the surface. The greater the water-vapor gradient, the more rapid is the evaporation and hence the greater is the heat loss. Cold, dry air overlying warm water therefore favors rapid evaporation. High winds increase evaporation by removing the water vapor.

The relative importance of the heat losses through evaporation and back radiation can be seen from the average heat budget between 70° S and 70° N, as follows:

humidity and cloud cover. With heavy, low-lying clouds present, the effective back radiation drops to less than 25 percent of that on a clear day, largely because the clouds are themselves sources of infrared and radiate heat into the ocean on their own account. Clouds prevent direct solar radiation from reaching the sea surface. Heat

	cal/cm²/min
Total heat received	0. 221
Evaporation losses	0. 118
Effective back radiation	0. 090
Conduction to atmosphere	0. 013
Total heat lost	0. 221

[2]*Oceanography for Meteorologists*. New York, Prentice-Hall, 1942.

MIXING PROCESSES

Convective Overturn

Thus far only the cooling effect of evaporation has been considered. When surface water cools, its density increases and causes convective over-turn. Equally important is the *increase in salinity* resulting from evaporation; the increased density arising from this cause contributes greatly to overturn and to the development of isothermal surface layers. Thus, cooling by evaporation is even less likely to be accompanied by positive temperature gradients than is cooling by back radiation.

Conditions that tend to lessen the salinity of the surface layer would, of course, have the opposite effect and would tend to favor the development of positive gradients. Such a condition might result from precipitation. For the ocean as a whole, however, evaporation exceeds precipitation. This fact is illustrated in figure 2-7. Shaded areas show regions where precipitation exceeds evaporation. The symbol 0/00 represents parts per thousand. Note that regions of excess evaporation in low latitudes and mid-latitudes correspond to regions of relatively high surface salinity and deep thermoclines. Just north of the equator and in

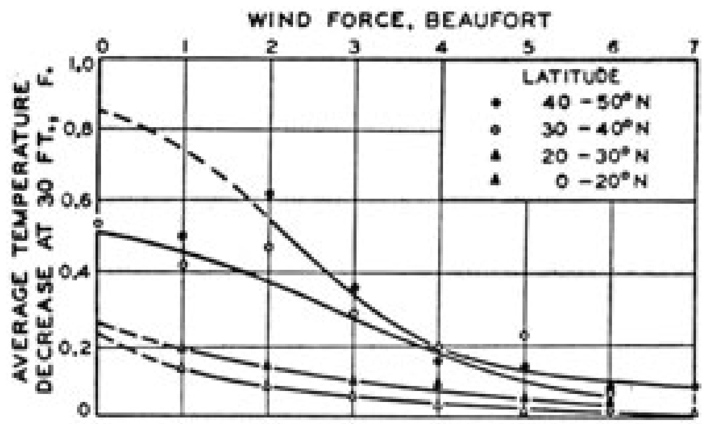


Figure 2-8.-Effect of wind on the temperature gradient in the surface layer at various latitudes.

The deficit in the water content of the ocean that is caused by the general excess of evaporation over precipitation is made up by run-off from land. Near land-especially near the mouths of rivers-surface salinities are lower than in the open ocean. This condition favors the development of positive temperature gradients, because it increases their stability.

Mechanical Mixing

Mechanical mixing is caused by wind and does not necessarily involve any gain or loss of heat. nevertheless, it may modify the temperature distribution. The effect of winds depends not only on their strength, but also on their duration and on the distance over which they have blown It is

latitudes above 40°, where precipitation exceeds evaporation, the surface salinity is low.

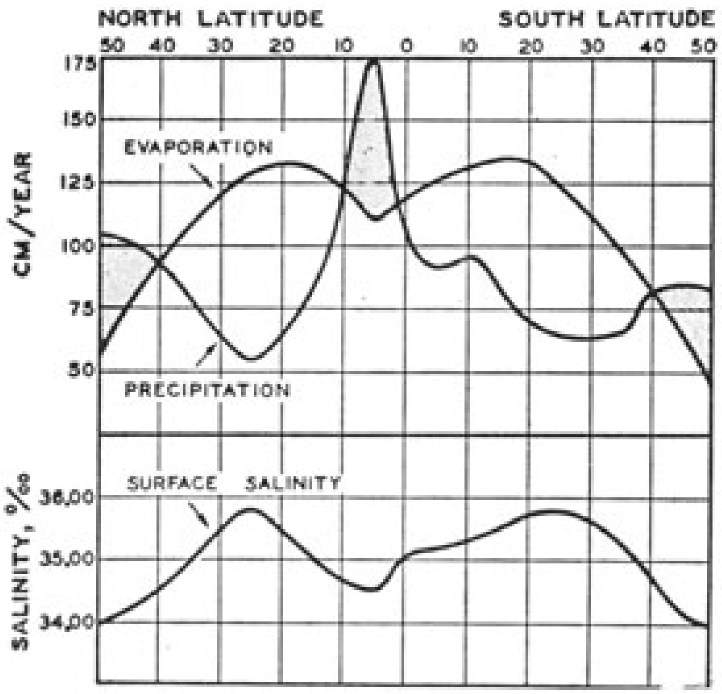


Figure 2-7 -Variation of average evaporation, precipitation, and salinity with latitude.

obvious that the first effect of the wind is confined to the immediate surface, but that the turbulence extends to greater depths after the wind has been blowing for some time. The original density distribution of the surface layer affects the rate at which the turbulence penetrates the layer. A very stable layer is less easily mixed.

Effect of Rotation of the Earth

The daily rotation of the earth about its axis also affects the depth to which the wind mixing penetrates. Present theories agree that a wind of given force ultimately produces a deeper mixed layer in low latitudes than in high.

This principle is probably part of the explanation of the data shown in figure 2-8, which indicate that strong negative gradients are most apt to be formed in high latitudes. If negative surface gradients are interpreted naively as being the result of solar heating alone this condition is most

---

unexpected, because heating is greatest at the equator. The necessity for considering all four of the major processes, with the detailed mechanisms causing them, is emphasized by figure 2-8.

CURRENTS

Drift Currents

The frictional drag of the wind sets up *drift currents* which flow at less than 3 percent of the wind velocity. These drift currents do not flow with the wind, but are deflected 45° to the right in the Northern Hemisphere and 45° to the left in the Southern Hemisphere. This deflection is caused by the earth's rotation and is closely related to the influence of the earth's rotation on the depth of mixing, which was just discussed.

density produces a temperature gradient such that in the Northern Hemisphere the water on the left side of the current has a lower average temperature than the water on the right side. This condition may be reflected by a thinner mixed layer or even by lower surface temperatures. In the Southern Hemisphere the structure is reversed.

Divergence and Convergence of Surface Currents

Divergence of the surface currents may occur under the influence of the wind. Examples of this effect are found along the western coasts of the continents and in the vicinity of the equator in the eastern parts of the Atlantic and Pacific. In these areas *upwelling* brings water toward the surface from moderate depths and the thermocline may be shallow or,

Permanent Currents

The redistribution of density resulting from the wind-drift currents in turn maintains the *permanent currents*. Under the influence of the steady wind systems, such as the trade winds in the low latitudes and the westerlies in high latitudes, these permanent currents form the large-scale current system of the oceans. They are thus partly the indirect result of geographic differences in the heating and cooling of the water and partly the result of wind action. The character of the currents is influenced also by the configuration of the oceans, but in general there are clockwise gyrals in the Northern Hemisphere and counter-clockwise gyrals in the Southern Hemisphere. Smaller currents related to land topography and local climate exist near the continents. A counter current flows eastward between the two westward-flowing equatorial currents.

The permanent currents have several effects on the temperature conditions. Currents with poleward flow tend to carry warm water into cooler regions. Conversely, currents with equatorward flow tend to carry cool water into warmer regions. Within the currents themselves the distribution of

rarely, absent.

The opposite effect, *convergence*, occurs in the center of the subtropical gyrals in the Northern and Southern Hemispheres. In these regions the surface water accumulates, and consequently the thermocline may be very deep.

Tidal Currents

Tidal currents in partially isolated, shallow areas have a marked effect on temperature conditions because they also cause turbulent mixing. In areas of strong tidal currents-for example in the English Channel-the water may remain virtually mixed throughout the year, although there is heating and cooling of the water column as a whole.

Internal Waves

Internal waves also affect the temperature distribution. The effect of these waves is reflected in a periodic rise and fall of the thermocline. Periods as long as 24 hours are known to exist, and recent studies have shown that periods of only a few minutes may occur. Whether there is a continuous spectrum of frequencies is not known.

Geographical Variations

DEPENDENCE OF THE ANNUAL CYCLE ON GEOGRAPHICAL LOCATION

The annual cycle in temperature conditions represents the net effect of the annual sequence in the various factors described, particularly in the amount of radiation received, the heat losses associated with evaporation, and the character of prevailing winds. In low latitudes where these factors do not vary appreciably there is little

change in conditions throughout the year, except that near the continental boundaries changing monsoon winds may introduce variable conditions.

The annual cycle is most conspicuous in the latitudes of from 40° to 50°. This condition is to be expected, because in these regions the surface experiences the greatest range of temperature. The effects of this great variation in temperature are magnified by the fact that in winter the cooling due to low temperatures is increased by the greater evaporation that occurs at this season. The resultant increase in the density of the surface water facilitates mixing and thus contributes to the seasonal variation.

The annual cycle is even more pronounced in regions near land, and in areas where heavy precipitation occurs and light winds prevail during the spring and summer. These conditions tend to induce even more extreme negative gradients than those shown in figure 2-2. These gradients can also be observed generally in areas of flow toward the equator, in which cool water is being heated-for example, off the California coast.

DEPENDENCE OF THE DIURNAL CYCLE ON GEOGRAPHICAL LOCATION

The diurnal change in temperature gradients is essentially similar in principle to the annual cycle, but the temperature changes are smaller and do not extend to such great depths. The incoming solar radiation depends on latitude, time of year, time of day, and cloudiness. The diurnal cycle of incoming radiation changes during the year, the variation being least near the equator and increasing toward the poles. Above the polar circles, of course, there are days of complete darkness during winter and continuous daylight during summer. The diurnal change is not necessarily cyclic, as is time annual change, and progressive heating or cooling of the water is characteristic in middle and high latitudes. Within the tropics, where the annual variation is small, diurnal changes are more nearly cyclic.

Even if the total heat absorption is the same, the character of the changes in temperature gradients may be quite different, because these changes depend on the previously existing gradients and on the wind conditions. A negative gradient near the surface is increased by incoming heat unless a strong wind (force 4 or greater) springs up. On time other hand, the changes in an initially mixed (isothermal) layer depend critically on the wind strength. The development of surface gradients is common when the wind force is 3 or less but is rare with winds above force 4. (See figures 2-3, 2-5, and 2-8.) In the trade-wind belts, therefore, development of surface gradients during the day is a rather rare condition-probably another factor to be considered in explaining figure 2-8.

Summary of Conditions for Temperature Gradients

The regional differences in temperature structure can be explained in terms of the factors described. The discussion can be summed up as follows:

An *isothermal layer near the surface* is the result of mixing. The factors inducing mixing are (1) wind, (2) radiative cooling, (3) evaporation, with its consequent cooling and salinity increase.

Strong *negative gradients* are the effect of heating a stable surface layer, without much wind mixing. Strong winds may tend to prevent the formation of negative gradients.

Positive gradients are produced only in areas where cool, dilute water flows or is formed on top of warm, more saline water. Measurable positive temperature gradients are most common during the fall and winter months in the northwestern Atlantic and Pacific Oceans, where cold, dilute coastal waters are driven offshore by the wind and flow over the warm. but saline ocean water of higher density.

Wakes

Echo formation from discontinuities in a medium, such as suspended air bubbles in water, has been discussed in chapter 1. In sonar, the principal source of discontinuities that produce echoes, reverberation, or scattering is the wake of a ship. The acoustic properties of a wake are important because of their influence on transmission and operational procedures.

42

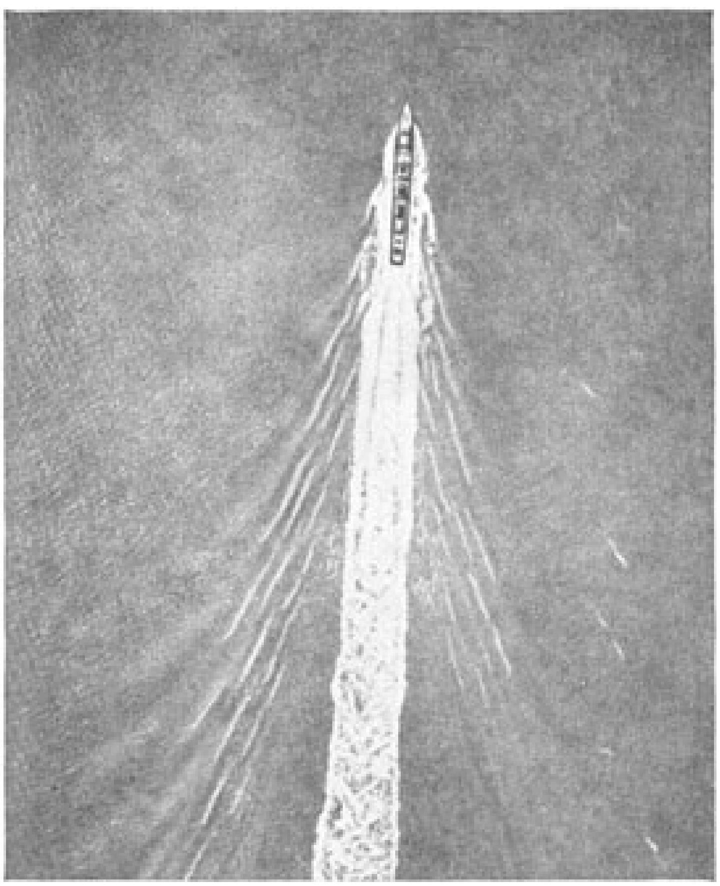


Figure 2-9. -Wake of U. S. S. Moole (DD) at 20 knots, From 2,500 Feet.

Figure 2-11. -Wake of surfaced submarine at 15 knots.

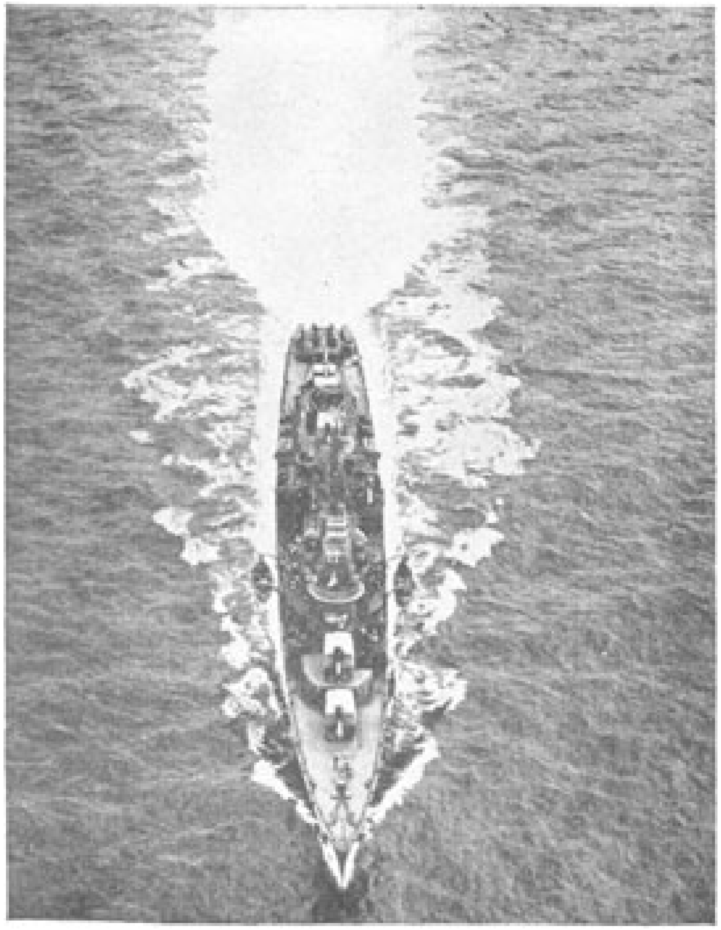


Figure 2-10. -Woke of U. S. S. Ringgold (DD) from 300 feet.

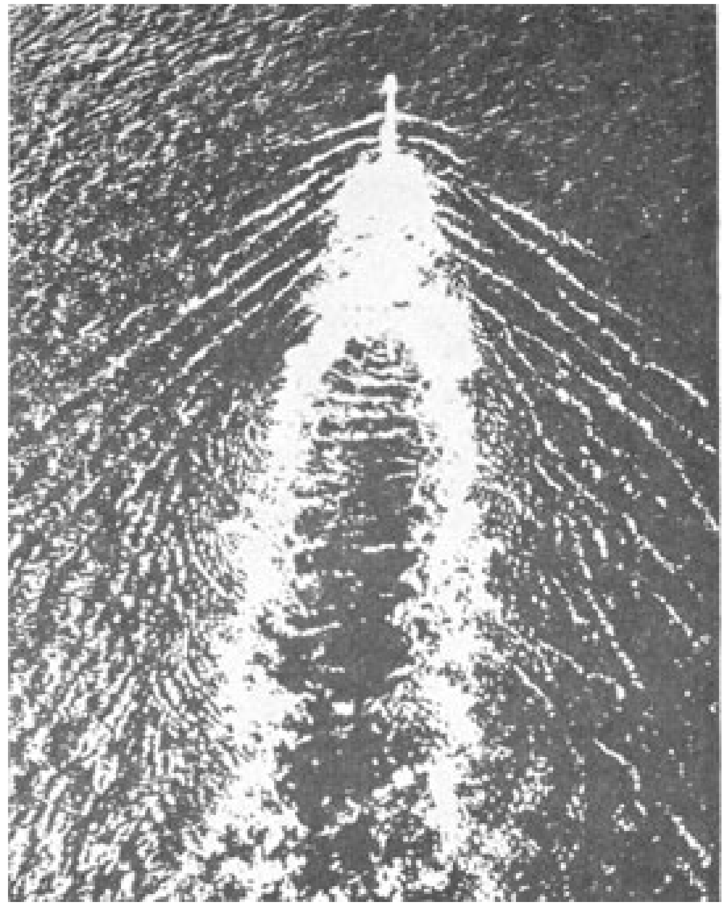


Figure 2-12. -Swirl behind submarine after crash dive.

VISUAL APPEARANCE

The wake of a ship is most readily seen from the air (figures 2-9 through 2-12). The surface waves that spread out in a V-shape behind the vessel and form a navigational hazard for nearby small craft are relatively inconspicuous from the air. Even the white bow wave, which breaks and sends foam back along the sides of the vessel, is inconspicuous compared to the wake of the turbulent, foamy water that fans out from the screws.

This turbulent wake spreads rapidly for a fraction of the ship's length, and thereafter widens only slightly. The divergence has been measured for the small sphere being recorded simultaneously with those from the wake.

Two general conclusions can be drawn from figure 2-13o(1) the wake echo gradually lengthens and becomes fainter, presumably because of the spreading of the turbulent wake and the gradual disappearance of the bubbles, and (2) the sphere

various wakes and found to vary from 0.5° to 5°. The foam, which makes it visible from a distance, gradually disappears, but not until long after the ship has passed. The visible wake of a high-speed vessel extends from 20 to as much as 50 ship lengths astern.

PHYSICAL PROPERTIES OF WAKES OF SURFACE VESSELS

It is fairly obvious that the violent disturbance which creates the turbulent wake gives it physical properties that differ to a greater or lesser extent from those of the undisturbed ocean surrounding it. For example, if there is a temperature gradient in the upper part of the ocean, the mixing of the surface water with that of lower layers gives the water in the wake a different temperature from that of the nearby water at the same depth. This effect has been observed by the use of sensitive recording thermometers. The mixing of water from different depths may also result in anomalous density gradients.

ACOUSTIC PROPERTIES OF WAKES

Of most interest at this point are the acoustic properties of the wake. They are probably all associated with the presence of entrained air bubbles. Aerial photographs show, that large numbers of bubbles remain in the wake for several minutes. It is likely that some remain suspended in the water even after the visible foam disappears.

These acoustic properties of the wake are easily demonstrated with sonar gear. Figure 2-13 shows a record of echoes obtained from the wake of the *E. W. Scripps*. This vessel ran between the echo-ranging vessel and a small sphere, the echoes from

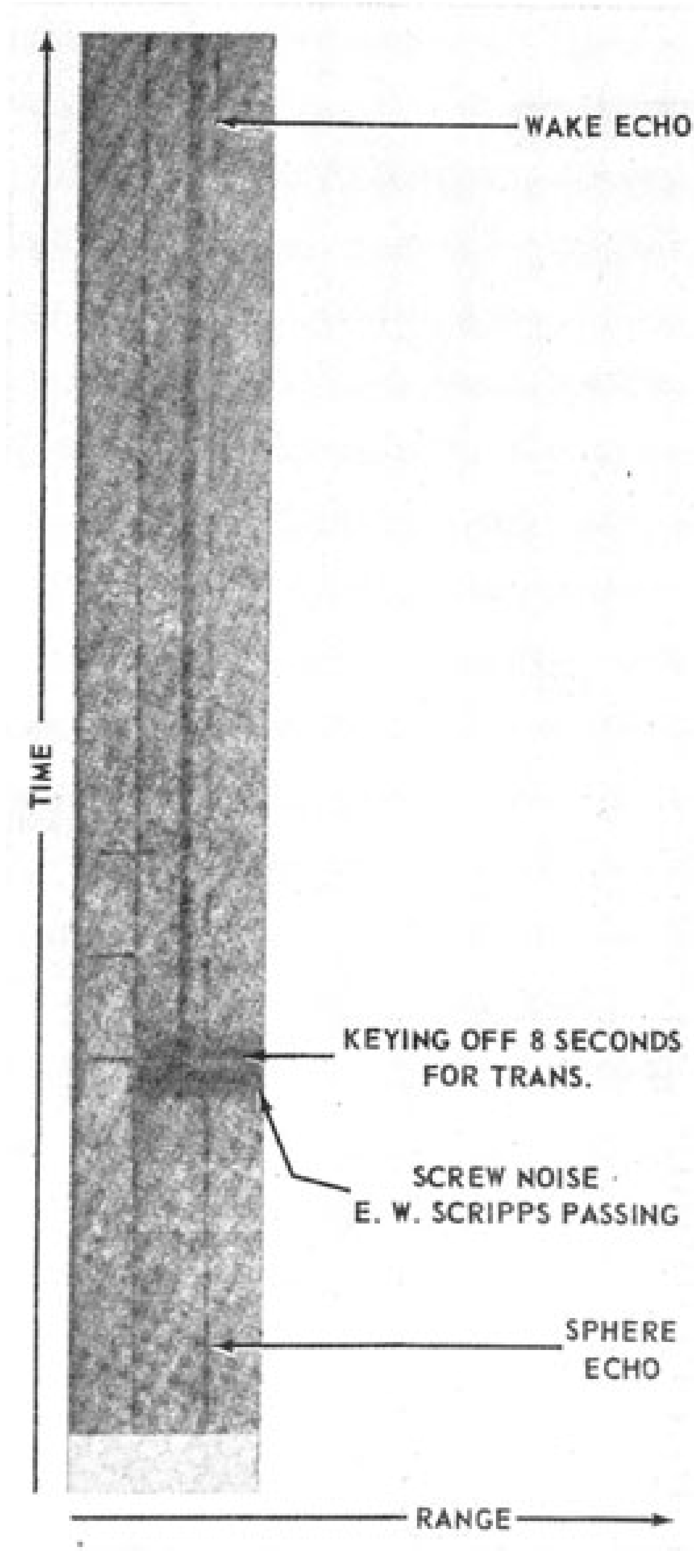


Figure 2-13 -Range recorder trace of wake echoes.

echo is weakened slightly but noticeably by the wake between the sonar and the sphere.

Thus, we may conclude that the wakes of surface vessels have two major acoustic properties-(1) they return echoes that are readily detectable by ordinary sonar gear, and (2) they act as acoustic screens, reducing the intensity of the echoes from targets on the far side of the wake.

Causes of Acoustic Properties of Wakes

The two most obvious differences between a surface wake and the undisturbed ocean are (1) the turbulence of the wake and (2) its content of bubbles. It is therefore reasonable to assume that one or both of these factors are the cause of its acoustic properties.

The possibility that turbulence is the cause of wake echoes is ruled out by theoretical considerations. It is true that when a sound wave passes through turbulent water it is scattered, but two facts exclude the possibility that this scattering is the cause of echoes-(1) the scattering from turbulence is very weak unless there are great differences in velocity between pairs of points separated by 1 wavelength of the sound, and (2) the intensity of the scattered sound depends strongly on the direction of scattering, and the intensity in the backward direction is zero. Thus, although turbulent water scatters sound, it does not return an echo.

Turbulence may be an indirect cause of the echoes by mixing the warmer surface water with that from below. From this mixing, irregular differences of temperature are produced, which cause irregular differences in sound velocity in the turbulent water. However, the magnitude of the expected effect is too small. To produce the observed echoes, temperature differences of nearly 1° F would have to occur between points that they disappear from the wake in a short time; their disappearance is also hastened by absorption of air by the sea water. On the other hand, echoes have been obtained from wakes more than 10 minutes after the vessel has passed, and there have been reports of echoes from wakes several hours old. The latter reports may be discounted, because it is very difficult to be certain of the position of a wake so long after the ship has passed, and it is quite possible that a school of fish, for example, might be mistaken for a wake under such circumstances. It is therefore necessary only to show that some bubbles remain suspended for periods of from 10 to 30 minutes.

Experimental evidence on this point was obtained by stirring the water of the pool at U. S. Naval Electronics Laboratory (USNEL) with an outboard motor. The acoustic properties of the water were studied with an echo sounder. It was found that sound was returned from the body of the water after stopping the motor. This return continued even after all the more obvious bubbles and turbulence had disappeared. Closer examination showed, however, that a relatively small number of small bubbles remained suspended. They were very difficult to see except when they drifted into a region of favorable illumination, so that neither their number nor their size could be accurately determined. It was concluded that sufficient bubbles were present to explain the observed effects. This conclusion was based on the consideration that very small bubbles are quite effective in scattering sound and rise very slowly.

The rate of rise of the bubbles which are most effective in scattering is about 1 yard per minute. These results for still water do not apply directly. to wakes or turbulent water. The long-lived bubbles observed in the USNEL pool did not show any marked tendency to rise but were carried in irregular paths by the motion of the water. This condition is analogous to the effect of air currents in keeping

only 1 wavelength apart. Such large temperature differences are very improbable. Moreover, if they were formed in some way, they would persist for a much longer time than wakes are observed to persist.

Thus, it may be concluded that the air bubbles in a wake are the major cause of the acoustic properties of the wake. Several objections have been urged against this conclusion. One objection is based on the supposed short life of bubbles in water. Bubbles rise to the surface and break, so

dust from settling. It is reasonable to suppose that the moderate turbulence in an old wake has this same effect and prevents the disappearance of the bubbles.

Propeller Cavitation as a Source of Bubbles

The second objection to air bubbles as the cause of acoustic properties in wakes is based on the fact that echoes are obtained from the wakes of

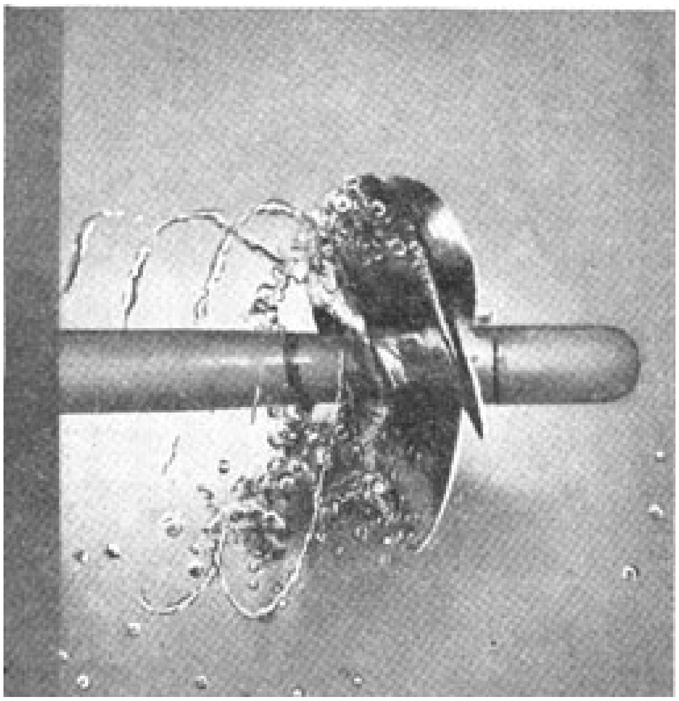


Figure 2-14 -Cavitating propeller.

submerged submarines and the idea that most of the bubbles in a wake come from the breaking bow wave. Aerial photographs strongly suggest that, this idea is not correct, because most of the foam appears to come directly from the screws. This idea is borne out by the observation that the wake laid by a vessel under sail is less acoustically active than the wake of the same vessel under power.

However, sea water is not pure. In the present discussion, dissolved air is the most important impurity. Dissolved air is present in such quantities that sea water boils at 60° F whenever the pressure is reduced much below 1 atmosphere. The bubbles produced by this boiling are filled principally with air, rather than water vapor. Once formed, these bubbles are apparently quite stable-that is, the rate at which the air is redissolved is very slow.

Even in the wakes of surface vessels, much of the foam is probably the result of cavitation, and only a part of it is probably caused by air dragged under from the atmosphere. In the wakes of submerged submarines the only sources of air other than cavitation might conceivably be leaky high-pressure air lines.

Dependence of Cavitation on Depth and Speed

Cavitation depends critically on propeller rpm. A given propeller at a given depth of submergence produces no bubbles unless its speed exceeds a certain critical value; when the speed exceeds N_o, the number of bubbles formed increases very rapidly, but not according to any known law.

Hence, most of the bubbles are caused probably by cavitation at the propellers. Photographs of this phenomenon are shown in figures 2-14 and 2-15. In figure 2-14, the water in the jet is moving away from the observer. The back of each blade is half covered with cavitation bubbles and a cavitation void which extends for some distance behind the blade, whereas the face of each blade is clean. In figure 2-15, the cavitation of the rotation of the propeller and the flow of the water in the jet from left to right gives a spiral pattern to the vortices.

The bubbles are formed far from the air-water interface and are not sucked under from the atmosphere. The mechanism of cavitation is apparently similar to that of boiling. Because of the motion of the screws the hydrostatic pressure is reduced; the boiling point of water is lowered by this reduced pressure and boiling occurs. For example, pure water boils at 60° F if the pressure is reduced much below one-sixtieth of an atmosphere.

The critical speed itself, however, depends in a simple manner on *h*, the depth of the propeller beneath the sea surface. Expressed as an equation, this dependence is

$$N_o^2/h = \text{constant.} \quad (2\text{-}1)$$

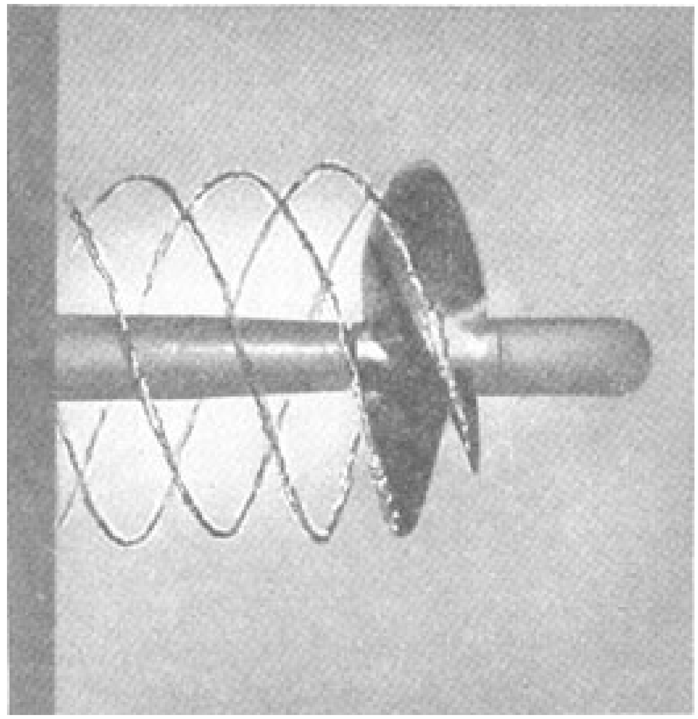


Figure 2-15 -Tip vortices emanating From a propeller.

Thus, if a given propeller begins to cavitate at 50 rpm when at a depth of 15 feet, it begins to cavitate at 100 rpm when at a depth of 60 feet and at 200 rpm when at a depth of 240 feet.

The constant in equation (2-1) depends on the design of the propeller, and on any accidental changes in its shape that may occur in service. A scratch or nick caused by some accident usually reduces appreciably the value of the critical speed. One remarkable property of cavitation is that the bubbles themselves scratch and scar the metal surface on which they are formed.

In *bubbly water* the average distance between neighboring bubbles is considerably greater than the average diameter of the bubbles but much less than the wavelength of the sound involved. For practical purposes, water may be considered to be bubbly if it contains less than 1 part per 1,000 (by volume) of air and foamy if it contains much more than this amount. The bubbles are *dispersed* if the average distance between neighbors is greater than both 1 wavelength of the sound and the average diameter. Thus, a portion of a wake may be dispersed for ultrasonic frequencies and bubbly for sonic frequencies.

This theory of the relation between cavitation and the acoustic properties of wakes has certain consequences that can be qualitatively checked. Thus, the wake of a submerged submarine should return echoes, but the echoes should be considerably weaker than when the ship is moving on the surface. They should also become progressively weaker as the depth of submergence increases. Finally, they should increase rapidly with propeller speed. All these conclusions are in general agreement with experience.

The propellers are probably not the only source of cavitation bubbles. Because the ship as a whole is moving through the water, cavitation can occur at other places. In general, the smaller the object, the lower is the critical speed at which cavitation occurs. Thus, small fittings or handrails on the deck of a submarine may become sources of cavitation bubbles when submerged.

PROPAGATION OF SOUND IN WATER CONTAINING BUBBLES

The theoretical discussion of the acoustic properties of water containing air bubbles is complicated, and the studies are not complete. To present the general ideas of the theory without confusion, it is convenient to introduce certain terms for the description of water containing bubbles.

In *foamy water* the average distance between neighboring bubbles is less than the average diameter of the bubbles. The walls separating the bubbles may occasionally be very thin, as with soap suds. The acoustic theory of foamy water has not been studied, but lack of this information is not important because wakes probably contain foamy water only at the air-water surface, where the bubbles tend to accumulate.

It would be useful to have information concerning the foamy, bubbly, and dispersed regions of typical wakes. Unfortunately, there is relatively little information of this sort other than that which can be obtained from the inspection of aerial photographs or deduced indirectly from acoustic measurements. The wake probably reaches the dispersed state between 5 and 10 ship lengths astern of the screws; it is foamy only in the immediate neighborhood of the screws or at the air-water surface.

Scattering and Absorption of Sound in Wakes

Except for some details, the theory of dispersed wakes is similar to the theory of reverberation.

Consider a region where the acoustic energy of a sound beam is flowing into a dispersed wake. The bubbles remove power from the beam at a rate that depends upon the intensity of the sound in the beam and the *total effective cross section* of the bubbles. Of the power removed from the beam, a fraction is reradiated as sound. The quantity of energy reradiated is determined by a factor called the *scattering cross section* of the bubbles.

The remainder of the power that is removed from the beam is converted into heat-that is, absorbed by the air of the bubbles and, to a lesser extent, by the water surrounding them. The quantity of energy converted into heat is determined by the *absorption cross section* of the bubbles. Thus the total effective cross section is a combination of scattering and absorption cross sections. Note that the total effective cross section determines the screening effect caused by a wake, whereas the strength of the wake echo is determined by the scattering cross section.

Interpretation of Scattering Experiments

Historically the study of scattering and absorption has played an important part in the development of various branches of physics. This development is especially evident in those branches dealing with radiations that are not perceptible by the unaided human senses-such as X-rays; α rays; β rays; γ rays; cosmic rays; and more recently, neutron rays. The scattering of visible light explains the color of the clear sky and other meteorological phenomena. The scattering of sound waves had not been studied in any systematic manner before World War II. During the war such studies were begun and are still far from complete.

The modern knowledge of the structure of matter, atoms, and nuclei is based largely on scattering experiments. Experiments on the scattering of sound and radio waves are unlikely to contribute much to this fundamental knowledge concerning the imperceptible structure of matter. Such experiments almost certainly will contribute much to the knowledge of the inaccessible parts of the ocean and the atmosphere. Thus studies of reverberation and of the scattering of sound by wakes are considered to be very important, even apart from immediate practical objectives.

The interpretation of the experiments has been the subject of much careful thought, and has resulted in many major advances in knowledge. However, examples of misinterpretations by conscientious and able experimenters are also numerous.

The most common error is the measurement of extraneous radiation along with radiation that is intended to be measured. In measuring the intensity of sound transmitted through a wake, it is equations to circumstances that do not conform to the assumptions made in deriving them.

TRANSMISSION OF SOUND THROUGH WAKES OF SURFACE VESSELS

A series of experiments on the wakes of destroyers and destroyer escorts was performed by the University of California, Division of War Research (UCDWR).[3] The procedure was as follows: One vessel carried a hydrophone and was dead in the water, while a destroyer ran past it on a straight course at a fixed speed. As soon as the destroyer had passed, a small launch got underway and carried the sound source from one side of the wake to the other. In this way it was possible to measure the intensity of the sound both when the wake intervened between source and receiver and when the source was on the same side of the wake as the receiver. After allowance was made for the difference in range when the source was on one side or the other of the wake, the apparent transmission loss caused by the wake was determined.

It is not certain that the result is free from error. In the first place, when the sound soured is on the far side of the wake, it is possible that some sound may pass under the wake and reach the hydrophone. This error was minimized by suspending both source and hydrophone about one-half the depth of the wake. In spite of this precaution, it must be emphasized that the values of transmission loss so obtained are possibly too low.

This source of error can be eliminated by making the measurement while the source is in the wake, but in that case the value obtained may be too large because of the effect of bubbles in reducing the output of the source. To some extent this value is counterbalanced because only part of the wake is

most important to shield the hydrophone from all sound that passes beneath the wake.

The interpretation of many laboratory experiments has been simplified by the use of opaque screens to shield the detector from extraneous radiation. Sometimes these screens have not been completely opaque. Often their edges have been the source of scattered radiation. In performing scattering experiments at sea, it is not possible to use such screens, so that the probability of extraneous sound is particularly great. Another error is the application of theoretical between the source and the receiver. The true value probably lies between the two measured values.

WAKE STRENGTH

Dependence on Age of Wake

Early experiments were performed with a single vessel, the U. S. S. *Jasper*, which ran on a straight

[3] *Sound Transmission Through Destroyer Wakes*, OEMsr-30, Service Project NS-141, Report M-189, UCDWR, March 8, 1944.

course, then circled and echo-ranged on its own wake.[4] These experiments showed that the level of the echo decreased fairly rapidly with the age of the wake. The results of various experiments ranged from 1.5 db per minute to 8 db per minute, with an average of about 4 db per minute. The levels of the echoes were compared with those of reverberation on the same day at the same range from the sonar. On one day this range was about 235 feet and the echoes were about 40 db higher than either volume or surface reverberation. These two kinds of reverberation were about equal at this range. On another occasion the range was 140 feet and the echo was 17 db higher than surface reverberation. A sea state *2* and wind force *3* prevailed on this occasion.

In view of the variability of reverberation from day to day, these observations have little absolute significance but serve to give some idea of the strength of echoes from the wake of a small slow-speed vessel. The values obtained for the rate of decrease of the wake echo have greater claim to validity and are in good agreement with other observations.

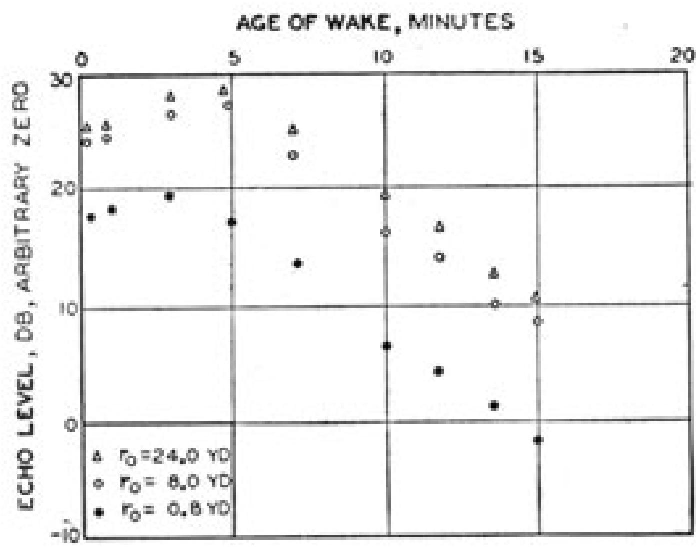


Figure 2-16 -Variation of the wake echo level with age of the wake, for various ping lengths at 24 kc.

maximum values until some time after the passage A the launch through the sound beam. Average values of the time of the maximum echo are shown in the second column of table 3. Thereafter, the echo intensity diminished at an average rate of about 7 db per minute for all three frequencies. The wake strength at the time of the maximum echo level was computed for each experiment, and average values are shown in the third column of table 3.

The difficulties inherent in performing experiments on wakes at sea led to an extended series of experiments in San Diego Harbor. A 40-foot motor launch was used to lay the wakes, which extended from the surface to a depth of about 5 yards.[5] There was some evidence that sound reflected from the bottom increased the strength of the echo. To minimize this effect, only echoes obtained at ranges of less than 100 yards are included in the following averages.

TABLE 3. -*Dependence of Wake Strength on Age of Wake*

Frequency (kc)	Time of maximum echo (sec)	Wake strength at time of maximum echo (db)
15	30	-2.9
24	50	+3.1
30	70	+8.4

Echoes were obtained by using 15-kc, 24-kc, and 30-kc sound. These echoes did not reach their

[4] Carl F. Eckart, *Echoes from Wakes*, NDRC C4-sr30-498. UCDWR, August 29, 1942. [5] Richard R. Carhart and George E. Duvall, *Acoustic Measurements of Surface Wakes in San Diego Harbor*, OSRD 1628, NDRC 6.1-sr 30-961, Report U-62. UCDWR, May 8, 1943.

Figure 2-16 gives further information concerning the behavior of wake echoes. The early period, during which the echo from the wake increases in level, is clearly evident, as is the later period during which the echo level decreases at a rate of about 1.8 db per minute.

Dependence on Ping Length

Figure 2-16 also brings out a dependence of echo level on ping length. The theoretical discussion has emphasized the analogy between wake echoes and reverberation. Essentially the wake is a part of the ocean from which the reverberation is especially high. If the ping length is shorter than the width of the wake the distinction between reverberation and wake echoes disappears. The number of scatterers returning echoes at any moment is determined, not by the extent of the wake but by the ping length.

Wake Strengths of Submarines

Many difficulties are encountered in experiments on the wakes of submerged submarines. The problems of navigation and seamanship involved in the

maneuvers are not always solved successfully, even by the ablest submariners. These practical difficulties and the low levels of the wake echoes account for the conflicting reports that have been made on the subject.

On one occasion echoes from the wake of an S-type submarine were recorded with standard echo-ranging gear operated at 24 kc. When this submarine was running at a depth of 45 feet, contact was maintained with the wake at a distance of 3,000 feet astern of the screws. At depths of 90 and 125 feet, the lengths of the contacts were 700 and 300 feet, respectively.

On a second occasion, an attempt was made to use a recording echo sounder for the study. This instrument had been successfully used in the study of the wakes of surface vessels. Consequently, it was mounted on a launch, and the fleet-type submarine ran on a straight course designed to carry it directly under the launch. This maneuver proved difficult to execute, but echoes from the hull of the submarine were obtained several times. The depth of the submarine varied from 65 to 200 feet. Echoes from the wake were never obtained at distances more than from 50 to 100 feet astern of the screws.

It had been hoped that this experiment would show whether the wake has a tendency to rise to the surface, as might be expected if bubbles are the primary cause of its acoustic activity. The results are inconclusive. It has been reported that, on several occasions, the wake of a submarine running at a depth of from 45 to 60 feet could be seen from the deck of a nearby surface vessel. This visibility was apparently due more to turbulence, which disturbed the surface, than to bubbles.

On a third occasion, 15 experiments were performed to measure the wake strength of a fleet-type submarine. Echoes from the wake could be recorded provided they were not more than 28 db below those from the submarine itself.

The operational problems were reduced to manageable proportions by the following procedure: The submarine started on the surface, running a course parallel to that of the echo-ranging vessel. The echo-ranging vessel ran at a slow speed, so that the submarine overtook it and passed through the sound beam while still on the surface and at a range of from 100 to 300 yards. About 90 seconds after passage the submarine dived rapidly to 90 feet and slowed down. Simultaneously the surface vessel increased speed and overtook the submerged submarine about 10 minutes later. It was found that these operations could be carried out satisfactorily except that it was difficult to adhere to the prearranged time schedule and that the submarine's submerged course often diverged appreciably from the course of the surface vessel. The timing of events was critical because of the limited supply of film in the magazine of the recording oscillograph.

Data recorded during such an experiment are summarized in figure 2-17. The lower half of the

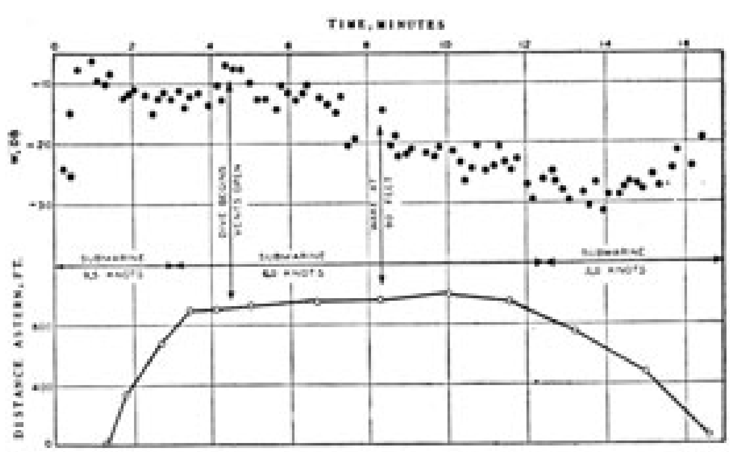


Figure 2-17 -Wake strength of a submarine.

figure shows the distance astern in feet. Note that the wake strength while the submarine was running on the surface was from -10 to -15 db. This wake strength was momentarily increased as the echo-ranging vessel passed the site of the dive, where the

type submarine running at various depths of from 45 to 400 feet. None of these experiments yielded echoes that were positively identified as caused by the wake, although echoes from the hull of the vessel were obtained. Some few echoes may have come from a short distance astern of the screws. Frequencies of 20 kc and 45 kc were used; 45-kc echoes from the wake would have been recorded provided they were not more than 14 db below those from the submarine itself. At 20 kc, the echoes from the wake would have been recorded venting of air from the ballast tanks presumably increased the bubble content of the wake. After the submarine reached the depth of 90 feet the wake strength varied between -20 and -30 db, even while the distance astern remained practically constant at about 900 feet.

TABLE 4 -*Wake Strengths of Submarines*

Submarine type	Frequency (kc)	Wake strength surfaced, 9 knots (db)	Wake strength submerged 6 knots (db)	Depth (ft)
S	60	-18	-26	90
S	45	-13	-24	90
Fleet	45	-13	-20	90
S	45		-33	45
S	20		-20	45

As the echo-ranging vessel overtook the submarine the wake strength again increased to -20 db.

The results of other experiments with submarines are listed in table 4. Ping lengths of from 8 to 24 yards were used in all the work summarized. Even when the submarine is running on the surface, the strength of its wake is very small. This fact can probably be explained by the low speeds at which the submarine moves.

Previous Chapter Sonar Home Page Next Chapter

CHAPTER 5

GENERAL REQUIREMENTS OF SONAR SYSTEMS

Operational Planning

This chapter discusses attempts that are being made (1) to cope with some of the special operational difficulties encountered in trying to obtain information by echo ranging and (2) to apply such information to tactical problems. From this information the military specifications of new equipments are obtained.

SEARCH OPERATIONS

Echo ranging is used by the Navy for several purposes, not all of which are directly connected with naval warfare. Echo ranging as an aid in antisubmarine warfare is only one of its applications, though perhaps it is the most important and dramatic. Regardless of its use, the success of echo ranging is conditioned by the systematic execution of a carefully considered operational plan. Such a plan is based on the consideration of the following functions that underwater echo ranging can successfully perform:

1. To establish contact with the target by using sound.
2. To maintain contact with the target and identify it.
3. To obtain accurate determinations of the *range* and *bearing* of the target.
4. To determine the rate at which the range and the bearing are changing-that is, the *range rate* and *bearing rate*.

Each of the last three of these functions successively depends on the preceding ones.

The present discussion is restricted to the

In a search operation three different missions can be assigned to the surface vessel or squadron. These missions are:

Hunt: To find as many enemy submarines as possible with little or no information as to their position at any earlier time.

Location: To find a specific enemy submarine whose position at an earlier time is known with reasonable accuracy.

Screen: To establish a zone (the screen) around a friendly area (a shipping lane or a moving convoy) so that all enemy submarines must pass through the screen in order to attack, and then to detect all enemy submarines while they are in the screen.

There are several differences among these three assignments. Hunt and location missions are offensive, and the submarine may be expected to use evasive maneuvers. The screen mission is defensive, and its objective-the prevention of a successful attack by a submarine-is partly achieved if the submarine is forced to use evasive tactics.

The success of these missions obviously depends on the probability of establishing sonar contact-that is, on the probability that when a ping is transmitted, a recognizable echo will be returned. Intelligent operational plans can be worked out only if all the factors affecting this probability are known and their effects evaluated. The effects of some of these factors are easily evaluated. For example, in the hunt operation, success may be equally probable if the echo-ranging vessel searches a wide area superficially or a smaller area intensively. In the location mission, success is assured if the vessel has sufficient speed to make an exhaustive

application of echo ranging to *search* operations in anti-submarine warfare as prosecuted by a surface vessel.

search of a sufficiently large but limited area,

provided that self-noise at high speed does not make the sonar inoperative.

The effects of other factors are not easy to evaluate. The most important factors to be evaluated are:

1. Range of the target.
2. Bearing deviation-that is, the difference between the actual bearing of the target and the transducer heading.
3. Relative bearing of the target.
4. Depth of the target.
5. Echo strength of the target.
6. Prevailing sound conditions.
7. Speed of the echo-ranging vessel.

To solve the problem of maneuvering a ship so as to bring the sonar into a position that will ensure a high probability of obtaining echoes, the cumulative effect of all the factors must be analyzed. Many operational rules, based on experience and a small amount of theoretical analysis, have been formulated. However, no complete solution of this operational problem has been made.

For this discussion, if it is assumed that adequate data are available on the last four factors listed, only the probability of establishing sonar contact based on the range and bearing of the target need be examined.

Probability of Detection

Single ping. -Assume that a target is in the vicinity of a sonar and that a single ping is transmitted. The detection probability can be exhibited on a contour map, like the one

The maximum value of a typical detection probability also should be large. In order to obtain a single number that describes the contour diagram, the areas between any two adjacent contours are multiplied by the average value of the detection probability, and the various products thus obtained are added. The result is called the *effective search area* of a single ping. For example, the area between the 30-percent contour and the 40-percent contour is measured, and this quantity is multiplied by 35 percent, the average probability in the area. Then the process is repeated for all the zones, and the sum of the individual products is computed.

In order to obtain a larger area, the beamwidth could be increased. An increased beamwidth, however, would make the bearing determination less accurate, and thus the gain of one advantage would cause the loss of another. In the design of an all-purpose pinging sonar the various requirements must be balanced carefully against one another.

shown in figure 5-1. This figure is entirely schematic and is presented merely to illustrate the discussion of general principles. It does not represent the facts of any actual situation.

The position of the echo-ranging sonar is indicated at the bottom of figure 5-1. If the target is situated on a given contour, the number shown on the contour designates the probability of detection. Such a number is called the *detection probability*. For example, if a target is on the 60-percent contour, a single ping will return a recognizable echo 60 percent of the time. If the target is inside the 60-percent contour, this probability will be greater.

For all search operations, it is important that the area of each contour be as large as possible.

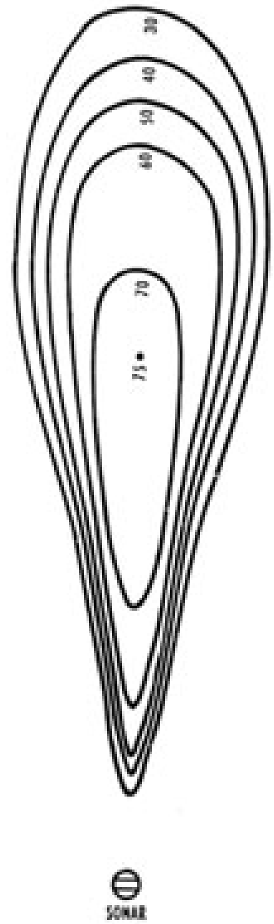


Figure 5-1. -Detection probability of a stationary target and a stationary sonar.

Successive pings.-In practice, surface vessels do not rely on a single ping for detection, although the tactical situation may force a submarine to do so. The analysis of the advantage of repeated pings in operational practice is complex; only a few major principles can be discussed here.

The simplest case is that in which both sonar and target are at rest and in which two pings are sent out. Then it is possible for an echo to be recognized (1) on both the pings, (2) on either of the pings, or (3) on neither of the pings.

Let W_1 be the probability that a single ping will return a recognizable echo for the given position of the target. Then the probability that the echo will *not* be detected is evidently

The probability that the second echo will not be detected is also

$1-W_1$.

The probability that neither of the two echoes will be detected is the product of the two probabilities-

$(1-W_1)^2$. (5-2)

Hence the probability that at least one of the two echoes will be detected is

$W_2=1-(1-W_1)^2$. (5-3)

If n pings are transmitted, the detection probability is

1-W_1. (5-1)

Let us assume that the detection probability for the second ping is the same as if the first ping had not been transmitted. This condition is not likely, for the operator may have been doubtful of the echo from the first ping and may have ignored it, but a doubtful echo from the second ping is, under these conditions, very likely to be considered certain-especially if a range recorder is used. This effect becomes increasingly important as the number of pings increases. For simplicity, however, memory and comparison effects are ignored.

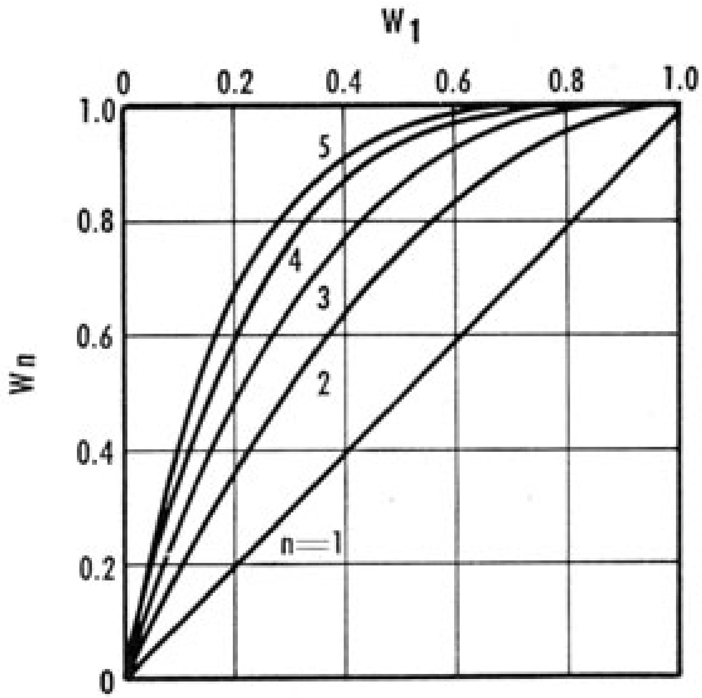


Figure 5-2.-Detection probability W_n for *n* pings in terms of the detection probability *W* of a single ping.

W_n= 1-(1- W_1)^n. (5-4)

Graphs of this equation for several values of n are shown in figure 5-2. Figure 5-2 shows an increase of detection probability with each successive ping. This increase is most rapid for intermediate values of W_1. If W_1>0.5, five pings will make detection practically certain.

Effects of motion. -If the echo-ranging vessel is in motion, the calculation of the probability of making sonar contact with a target by using successive pings becomes more complicated. If the target also is in motion, additional complications arise.

In the case of a stationary target and a moving echo-ranging vessel, suppose that the target was on contour *W*' of the first ping but that the motion of the sonar has resulted in placing it on contour *W*" of the second ping. Then, by reasoning similar to that in previous paragraphs and again i by ignoring memory and comparison effects, the probability of detection by either or both of the two pings is

W=1-(1-*W*')(1-*W*"). (5-5)

Values of this function are given in table 8.

TABLE 8.-*Detection Probability for Two Pings-Moving Sonar and a Stationary Target*

W"\W'	0.1	0.2	0.3	0.4	0.5	0.6	0.7	0.8	0.9
0.1	0.19	0.28	0.37	0.46	0.55	0.64	0.73	0.82	0.91
.2		.36	.44	.52	.60	.68	.76	.84	.92
.3			.51	.58	.65	.72	.79	.86	.93
.4				.64	.70	.76	.82	.88	.94
.5					.75	.80	.85	.90	.95
.6						.84	.88	.92	.96
.7							.91	.94	.97
.8								.96	.98
.9									.99

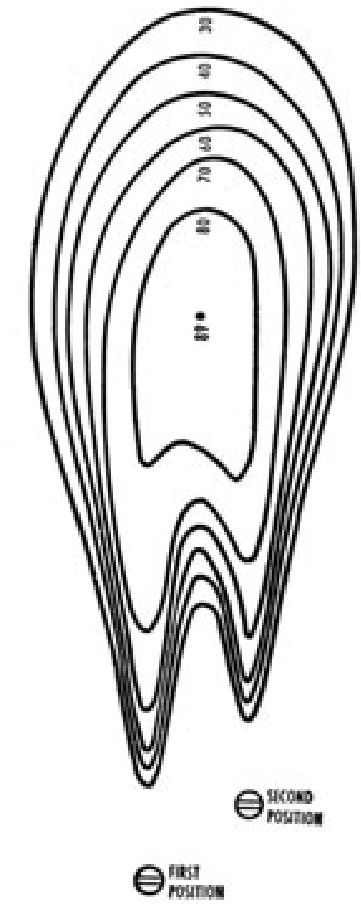


Figure 5-3.-Detection probability of a stationary target and a moving sonar.

Arbitrary values of *W*' are arranged in the top row, those of *W*" are in the left-hand column, and the corresponding values of W are in the body of the table. For example, suppose that when the first ping is transmitted, the target is on the 60-percent contour and that the motion of the sonar has resulted in placing it on the 50-percent contour for the second ping. Then *W*' equals 0.6, *W*" equals 0.5, and from the table, *W* equals 0.8.

Table 1 can be used to construct a contour map similar to that in figure 5-1. Such a map is shown in figure 5-3. The two successive positions of the sonar are shown at the bottom. It is assumed that the detection probability of each ping is identical with that diagrammed in figure 5-1, and that the pings were transmitted with the same transducer heading. The motion of the transducer between Moreover, the maximum value of the detection probability has increased from 75 percent for the single ping to nearly 90 percent for the two pings. Consequently, the effective search area of the two pings is more than doubled.

The amount by which the effective search area of the overlapping pair exceeds twice the area of a single ping has been exaggerated by the exaggerated motion of the sonar. In practice this amount is somewhat less than that shown, but the effect is nevertheless appreciable. Also, in practice, more than two overlapping pings are used, and the probability of echoes is increased further.

A moving target has a different effect than a moving sonar. Suppose that a moving target is detected at a certain point, *P*, at a time, t_o, and that at a later time, *t*, it is necessary to estimate

pings has been greatly exaggerated for purposes of illustration.

Comparison with figure 5-1 shows that each contour-for example, the 50-percent contour-has greatly expanded and encloses more than twice the area of the same contour for a single ping.

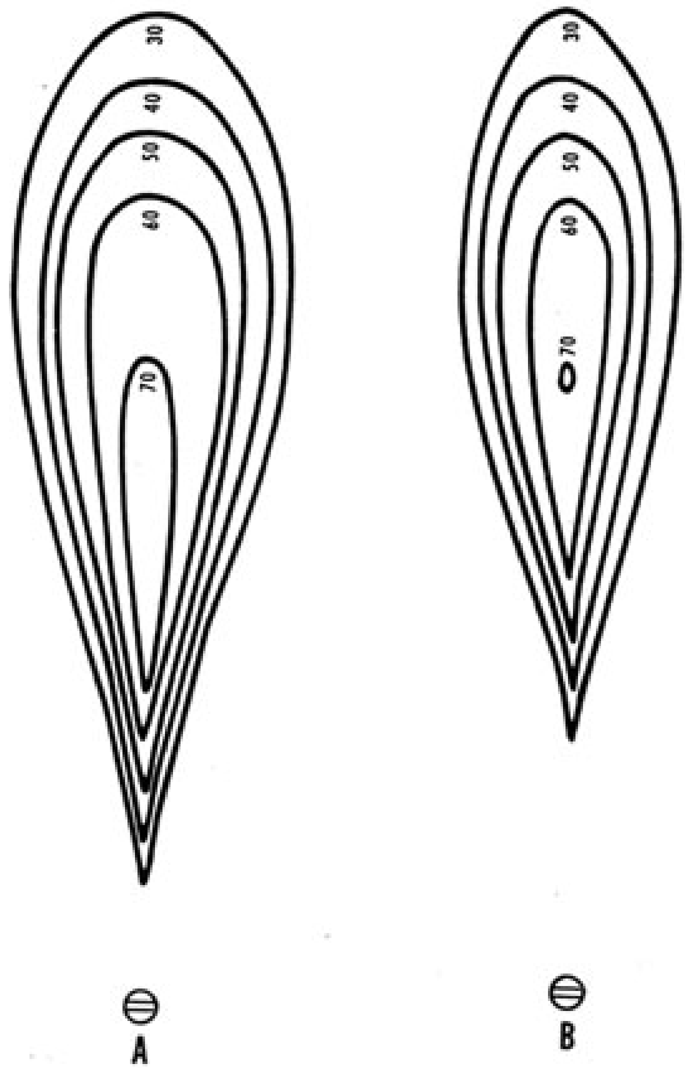


Figure 5-4.-Detection probability of a moving target and a stationary sonar. A, Change brought about in figure 5-1 after an interval of time; B, change in figure 5-1 after twice the interval.

given speed. As the time interval, t minus t_o, increases, these radii also increase because the unknown motion of the target has more time in which to take effect.

These same considerations can be applied to the time interval between pings. If the ping is sent out at time t_o, figure 5-1 shows the probability that, if the target is at a given place, it will detected. At a later time, t, but before the next ping, the target may have moved. Consequently, figure 5-1 does not show the probability that, if the target is at a given place at this

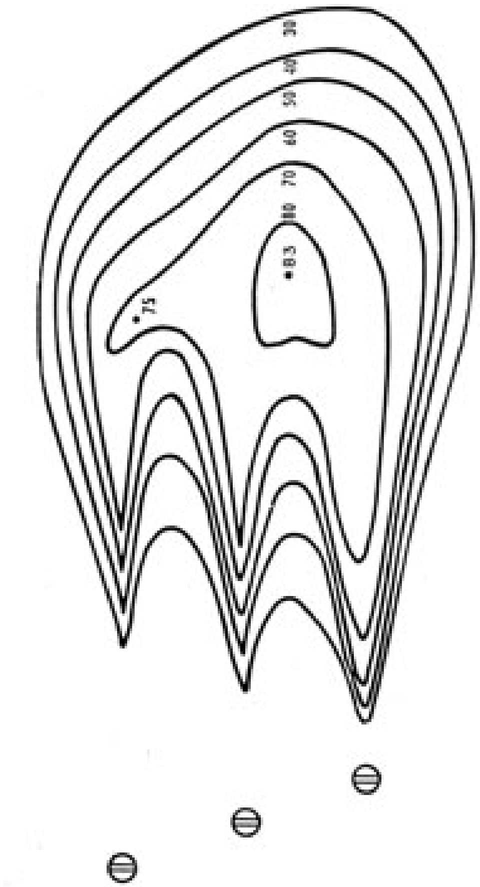


Figure 5-5. -Probability contours for three successive pings, allowing for the motion of both target and echo-ranging vessel.

its position. In order to illustrate the principles involved, suppose that between t_o and t no further pings are sent out and that the direction and speed of the target are unknown. Then it is possible to draw probability contours, showing the probability of the locations of the target at given points at time *t*. These contours are circles with centers at point *P*. The radii of the contours depend on the probability that the target moves with the

later time, it would have been detected at the earlier time t_o. But it is possible in principle to work out the contours for this "prior-detection" probability. The unknown motion of the target causes the contours of high probability to shrink as *t* increases. This effect is shown schematically in figure 5-4, A and B, for two successive values of *t*.

If several successive pings are sent out, shrunken prior-detection contours must be combined, as explained with figure 5-3. The result of such a succession of pings is shown schematically in figure 5-5. This figure represents the state of affairs at the time the echoes from the third ping are being received, and the contours show the probability that, if the target is then at a given point, it will be detected then or will have been detected earlier.

The motion of the sonar and target is exaggerated in figure 5-5 to emphasize important points. Note that, because of the unknown motion of the target, the 80-percent contour in figure 5-5 has a much smaller area than the 80-percent contour in figure 5-3. This condition exists even though the contour in figure 5-3 is based on three pings and that in figure 5-5 on only two.

Target Bearing

The foregoing considerations of the probability of establishing sonar contact have been restricted to simple conditions. In general, the possibility of taking action against a target in a given area depends on (1) how completely the area can be searched in a short time and (2) the ability of the operator to maintain sonar con tact with the target once he has contacted it. The first of these requirements makes it desirable to design the sonar so that the search area of the ping is large. The second requirement conflicts with the first. Special devices have been designed to satisfy these conflicting requirements. These devices will be discussed after a preliminary examination of the operational problem in terms of the simplest sonar.

MAINTAINING CONTACT

After a signal has been transmitted, the sonar operator is on the alert for a sound contact with the target-either as he listens to the sound from

the loudspeaker or as he watches the chemical range recorder. For example he might detect a break in the background reverberation or noise.

Having made a contact, he is concerned chiefly With maintaining it. Maintaining contact is difficult with ordinary sonar gear. The target may move out of the sound beam, either to the right or to the left. Because of the relatively long interval between echoes, the uncertainty as to the direction in which the beam should be rotated is a serious shortcoming in sonar design.

BEARING DEVIATION

The target bearing is the direction of the line joining the transducer to the center of the target and is not necessarily given by the *transducer heading*, which is the direction of the axis of the sound beam. Because of the width of the sound beam, an echo may be received even when the axis does not bear on the center of the target (for example, the conning tower in a submarine). Thus, the target bearing and transducer heading may not coincide. The difference between them is called the *bearing deviation*. When the bearing deviation becomes greater than a certain amount, the echoes become too weak to be heard.

As the sonar operator has control of the transducer, he knows its heading. The conning the sound beam is trained off the target, the motion of the transducer is reversed and is continued until the sound beam leaves the target on the other side. This method tends to eliminate bearing uncertainty. Whenever no echo is obtained the operator knows on which side of the beam he will find the target. The two limiting transducer headings thus obtained are called *cut-ons*. The average of two successive cut-ons is taken as the best approximation to the target bearing.

Although the procedure is practicable, it has many disadvantages. It is time-consuming, for it requires at least four, and often more, pings to obtain one value of the target bearing; hence, before this value is known to the sonar operator, the target may have moved, rendering the information more or less obsolete.

SPLIT TRANSDUCERS

Present solutions of maintaining contact and determining bearing all involve the use of a transducer that has been split into two or more segments. The first application of the split transducer was with searchlight equipment. The two hydrophones were constructed in semicircular shape and of such dimensions that they could be mounted in the same space as the older circular transducers. Moreover, if electric connections are changed before transmission, the projected sound beam can be made

officer, however, wishes to know the target bearing. If the bearing deviation is small, it can be ignored. Unfortunately, every attempt to reduce it increases (1) the probability that the target may move out of the sound beam and (2) the seriousness of the uncertainty mentioned above. Thus, every solution must be a compromise between conflicting requirements.

It is not only the beam pattern of the transducer and the target width that affect the possible magnitude of the bearing deviation; the echo level and the level of the background noise and reverberation are also instrumental. If reverberation is limiting, the possible deviation also depends on the Doppler shift of the echo.

CROSSING THE TARGET

The first solution of maintaining contact and determining bearing was the operation known as *crossing the target*. In this operation, the transducer heading is systematically changed more rapidly than the target bearing changes. When

identical with that of the older circular transducer.

The physical principles involved can most easily be explained by considering a pair of identical hydrophones, mounted a distance a apart, with

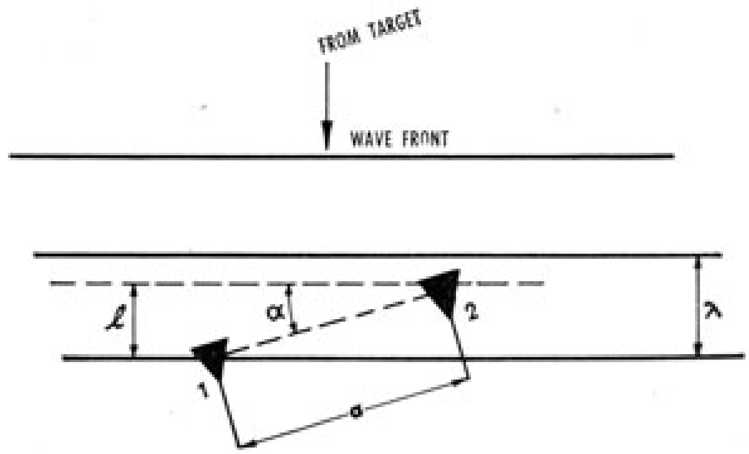


Figure 5-6. -Three successive stages in the passage of a plane wavefront from the target to a transducer having two hydrophones (marked "1" and "2") spaced a distance a apart.

phase angle β can be calculated as follows: After reaching hydrophone No. 2, the wave must travel a distance, *l*, before reaching No. 1. This distance is

$l = a \sin \alpha,$ (5-6)

which is l/λ wavelengths. Because 1 wavelength is equivalent to a phase change of 360°, the angle β is

$\beta = 360° (a/\lambda) \sin \alpha.$ (5-7)

If the current generated by No. 1 is

$C_1 = C(\alpha) \cos \omega t,$ (5-8)

then that generated by No. 2 is

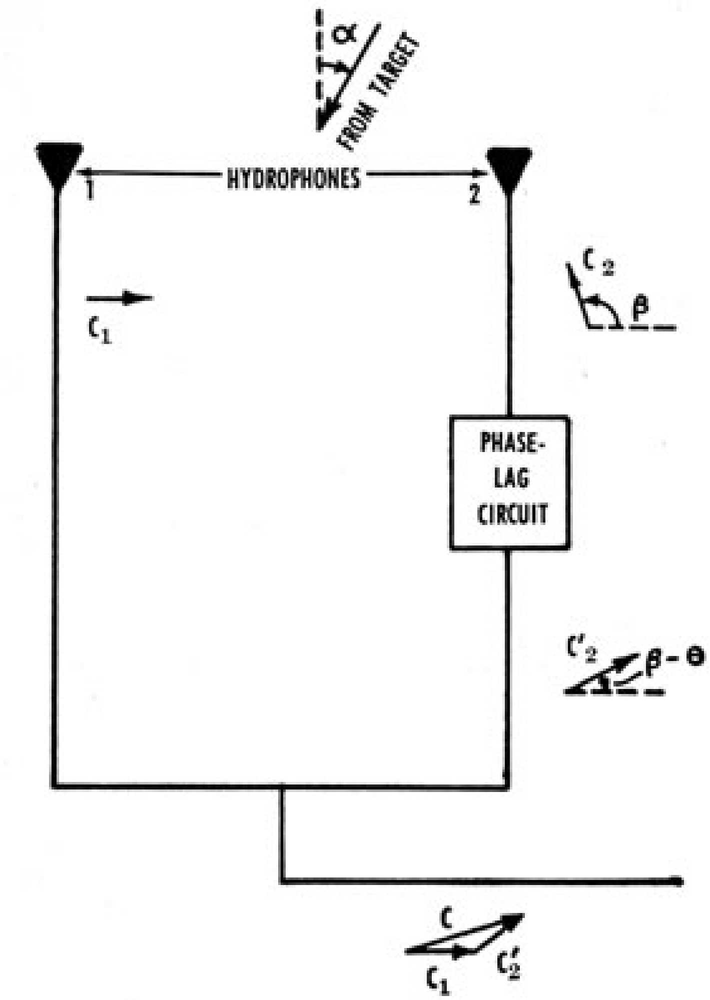


Figure 5-7. -Diagram containing phase-lag circuit, showing how a desired phase difference between the currents from the two hydrophones is obtained.

their acoustic axes parallel to each other and perpendicular to the line joining the two hydrophones. The general arrangement is shown schematically in figures 5-6 and 5-7. It is assumed that the pattern of the two hydrophones consists of a single broad lobe, as shown by the dotted line of figure 5-9.

Suppose an echo or other single-frequency sound is incident on the hydrophones from a direction that makes the angle α with the acoustic axes. Each wave then reaches the hydrophone closest to the target before it reaches the other hydrophone, and the alternating currents generated by them are not in phase. Under the circumstances shown in figure 5-6, the current from No. 2 is in advance of

$C_2=C(\alpha)\cos(\omega t+\beta)$. (5-9)

The function $C(\alpha)$ is determined by the directivity pattern of the separate hydrophones-shown by the dotted curve in figure 5-9. The graphs of the two currents, C_1 and C_2, are shown in curves *A* and *B* of figure 5-8.

If the current from hydrophone No. 2 is passed through a phase-shifting network, the phase shift β can be altered by any desired amount-say θ.

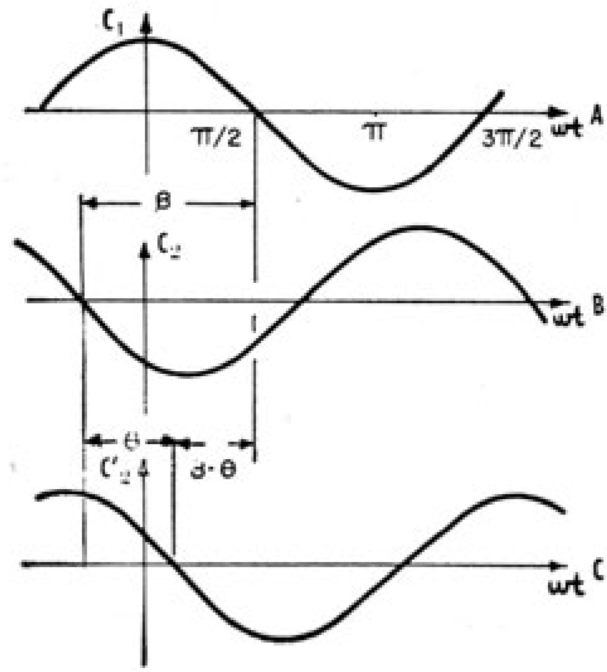


Figure 5-8. -Currents C_1, C_2, and C_2', of figure 5-7, plotted against the phase angle ω*t*, and showing the phase difference β and β-θ of figure 5-7.

that from No. 1. This condition is shown in curves *A* and *B* of figure 5-8. The

---

The result is the current

$C_2' = C(\alpha) \cos(\omega t + \beta - \theta)$, (5-10)

which is shown graphically in curve *C* of figure 5-8. The vector diagrams of the circuit shown in figure 5-7 indicate the relation of the three currents. If C_1 and C_2' are combined, the resulting current[1] is

$C = C_1 + C_2' = C(\alpha)[\cos(\omega t) + \cos(\omega t + \beta - \theta)]$ (5-11)

$C_1 + C_2' = 2C(\alpha) \cos\frac{1}{2}(\beta - \theta) \cos[\omega t + \frac{1}{2}(\beta - \omega)]$ (5-12)

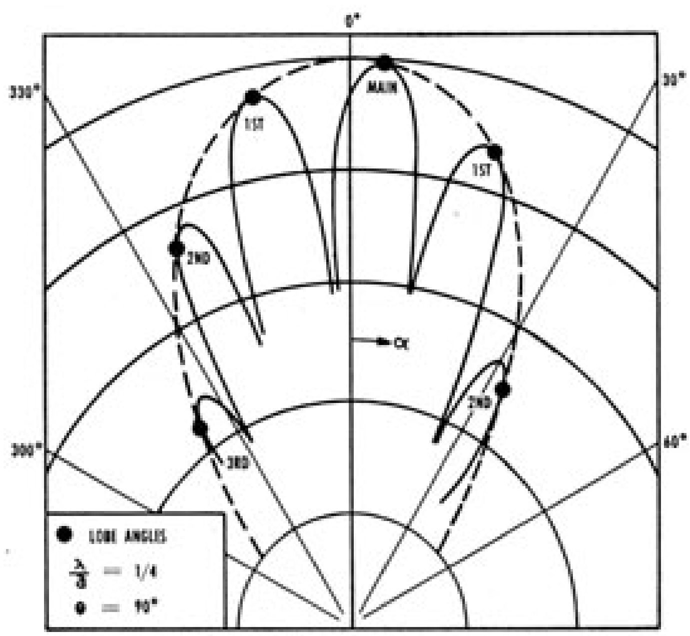


Figure 5-9. -Graph of equation (5-13) for *a*/λ=4, and for θ=90°.

The level of the electrical output is thus

$L = 20 \log [2C(\alpha)] + 20 \log \cos [\frac{1}{2}(\beta - \theta)]$. (5-13)

The first term of this expression is essentially the directivity pattern of the individual hydrophones.

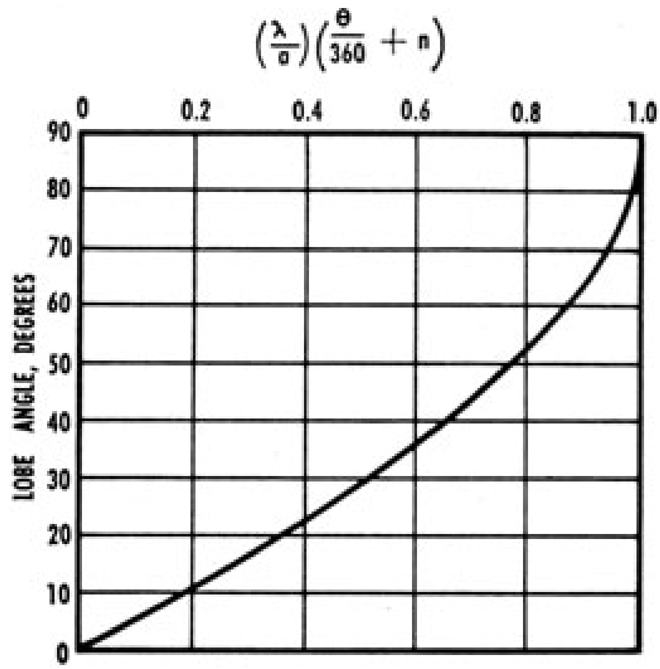


Figure 5-10. -Values of the lobe angles, shown in figure 5-9, as a function of λ/*a* and *n*, the order of the lobe.

As a result of the phase-shifting network, the axis of the new main lobe does not coincide with that of the original lobe, and the side lobes are not symmetrically located. Figure 5-10 can be used to determine the positions of the lobes for any value of the quantities θ, λ/*a*, and the order of the lobe, *n*.

In this graph (figure 5-10) the lobe angle is the point at which the new and original beam patterns are tangent (figure 5-9), and the integer *n* is zero for the main lobe, ±1 for the first lobes on each side, ±2 for the pair of second lobes, and so forth. The phase lag, θ, is measured in degrees.

The second term also depends on the direction from which sound comes, because β depends on α.

The graph of the resultant level, *L*, for the case in which is *a*/λ is 4 and θ is 90°, is shown by the solid line of figure 5-9. As a result of connecting the two hydrophones together, the single broad lobe of each obviously has been changed into several narrower lobes.

---

[1] In deriving the equation for *C*, the following trigonometric relation has been used: cos *A*+cos *B*=2 cos ½ (*A*+*B*) cos ½ (*A*-*B*).

239276°-53-8

Figure 5-11. -Shift of main lobe of beam pattern in the BDI system.

Bearing-Deviation Indication

All bearing-deviation indicating (BDI) devices use split transducers. The purpose of BDI devices is to translate into a polarized-magnitude difference the small echo-signal phase difference between the two halves of the transducer.

For transmission the two semicircular parts are connected so as to produce the normal beam of a

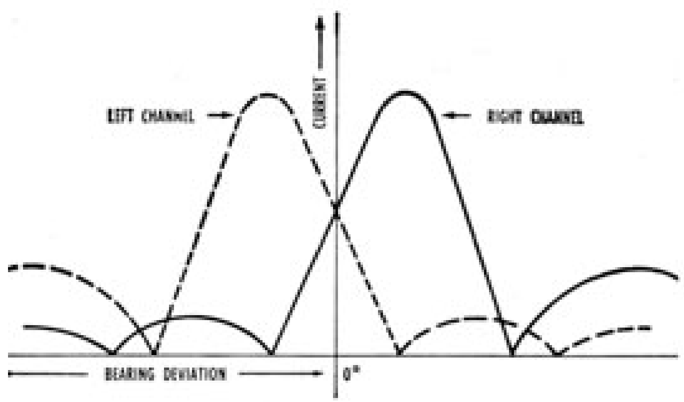


Figure 5-13. -Graph of currents from the two channels as a function of bearing deviation.

in figure 5-7. The connections of the left channel differ only in that the phase lag also is introduced into the output of No. 1. The beam pattern for the right channel thus has its main lobe deflected to the right, as shown by the right-hand curve of figure 5-11; the main lobe of the left channel is deflected to the left. These deflections are shown more clearly in the rectangular-coordinate system used in figure 5-13. The ordinates are the currents out of the two channels. In practice these currents are rectified, as indicated in figure 5-13; the diode rectifiers are shown in figure 5-12.

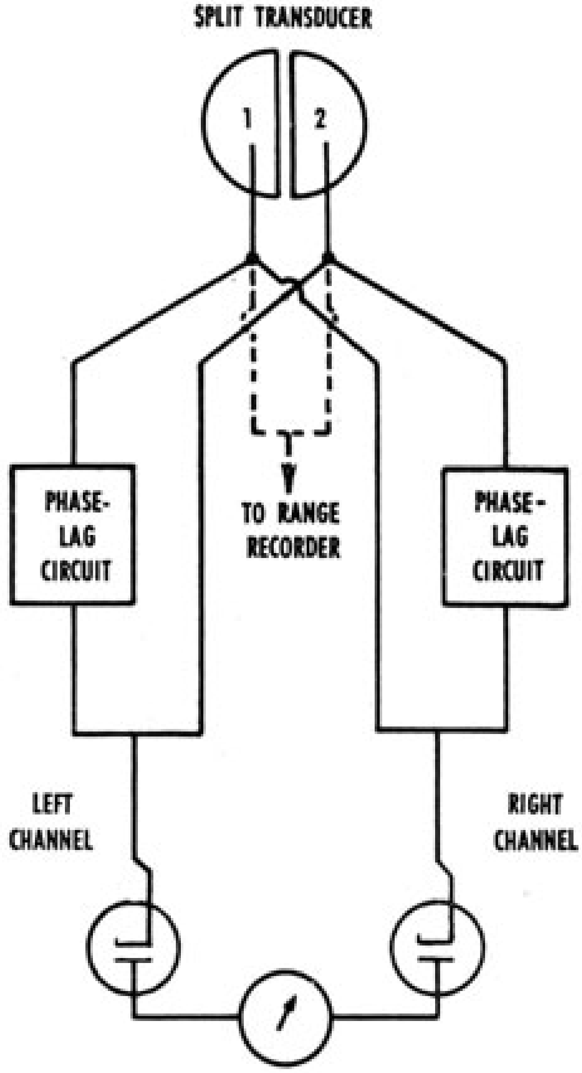


Figure 5-12. -Diagram of the BDI system.

circular diaphragm, which is illustrated by the center pattern of figure 5-11. The center pattern shows the normal beam of a circular diaphragm. The two side patterns show the beams for the two halves of the circuit. For reception, the two halves are connected as shown in figure 5-12.

Note that there are two symmetrical output channels. The connections of the right channel are the same as those for the pair of hydrophones

The rectified output currents may be used for various purposes. They are commonly connected to an indicator, which may be a cathode-ray oscilloscope, in such a way that the deflection of the

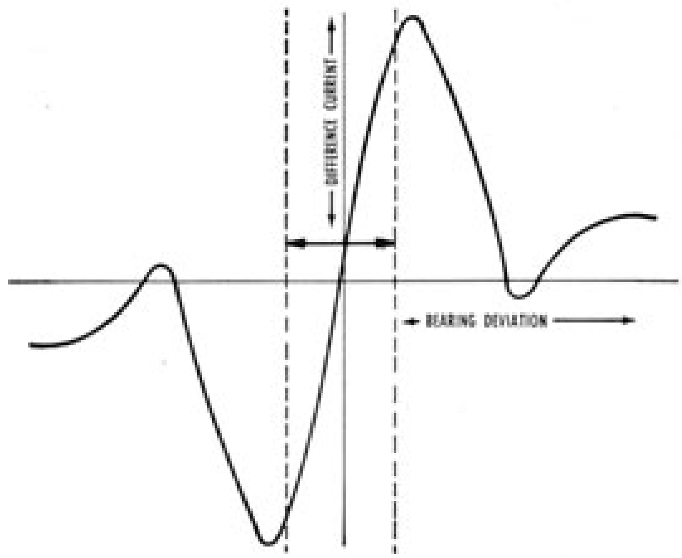


Figure 5-14. -Graph of the difference between the currents from the two channels as a function of bearing deviation.

indicator is proportional to the difference between the currents in the two channels. This difference is plotted as a function of bearing deviation in figure 5-14. Note that if the bearing deviation is not too great, the difference current is proportional to it. Confusion can occur if the deviation is greater than the limits set by the double arrow of figure 5-14.

Standard Bearing-Deviation Indicator

The standard BDI provides a visual indication of the sound incident on the transducer. When the transducer is trained on the exact center of the source (figure 5-15, B), the incident sound strikes both halves of the diaphragm simultaneously. This condition is indicated by a brightening of the luminous trace on the screen of the cathode-ray tube.

When the transducer is trained slightly off the center of the target (figure 5-15, A and C), the incident sound waves strike one of the transducer halves before the other. This action causes the brightened spot on the screen to be deflected in the direction of the half on which the sound first impinges. A deflection to the left thus would show (1) that the source is to the left of the transducer bearing and (2) that the operator must train left to get a center bearing. Conversely, a deflection of the brightened spot to the right would indicate that the operator must train right. A strong signal can produce a right and a left deflection of equal magnitude, thus indicating a center bearing.

Because the BDI reacts to all sound energy incident on the transducer, it must be used in conjunction with a loudspeaker in order to distinguish between echoes and reverberations (figure 5-15, D), and between echoes themselves-particularly between the echo from a submarine and that from its wake. For this purpose the visual perception is supplemented by listening to a loudspeaker.

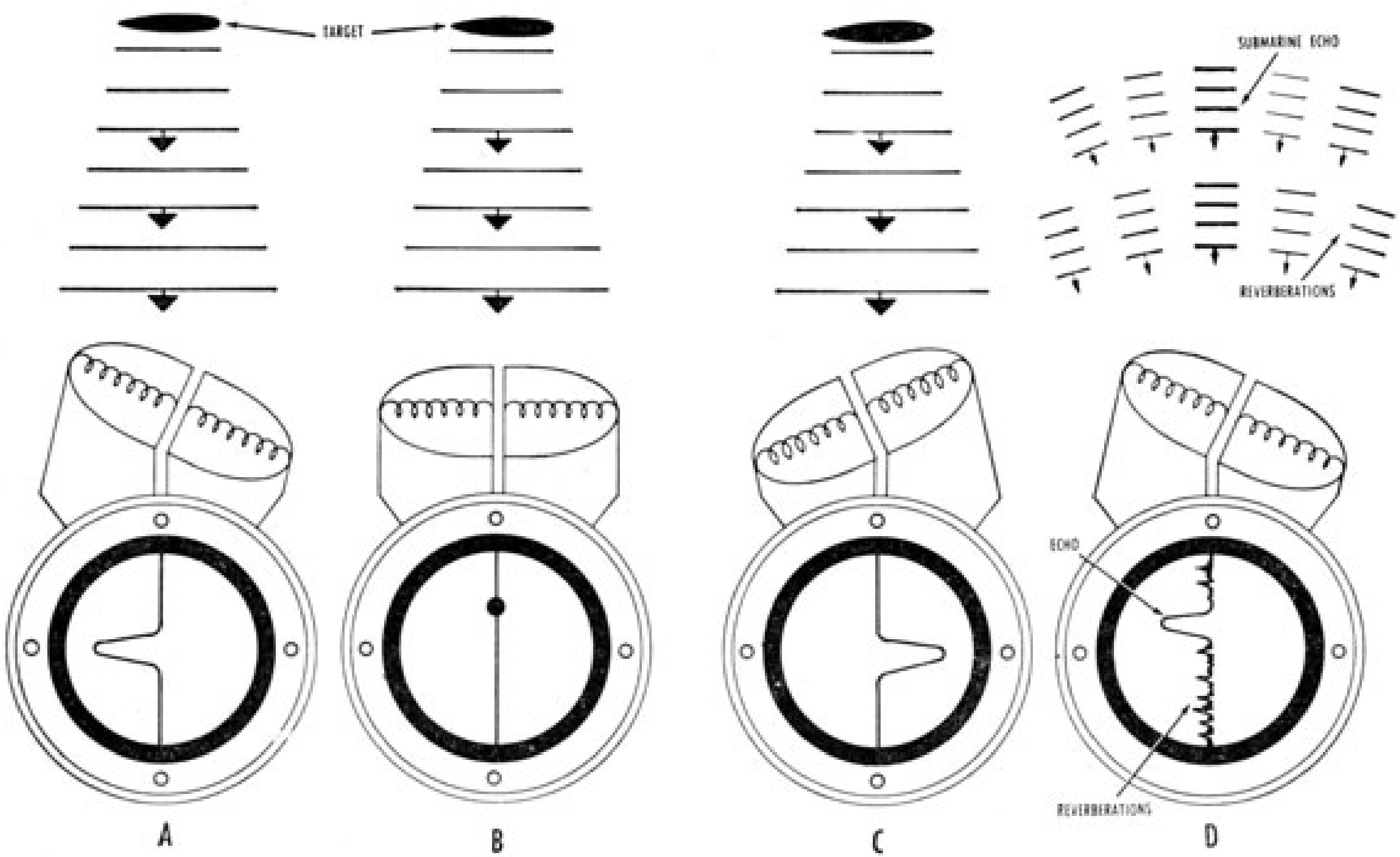


Figure 5-15. -Diagrams illustrating BDI. A, Deflection of the trace to the left by a target on the left of the transducer heading; B, transducer heading on the center bearing, causing the trace to brighten; C, deflection of the trace to the right by a target on the right of the transducer heading; D, echo distinguished from reverberations.

Scanning Sonar

The problem of rapidly searching a wide area led to the development of scanning sonars. The principle employed is to use the necessary interval between pings to search the widest possible area. In this way the area searched per ping and the amount of information received per unit of time are both increased. Two main types of scanning sonar have been designed-one transmits short pulses of sound, and the other transmits a continuous signal of varying frequency.

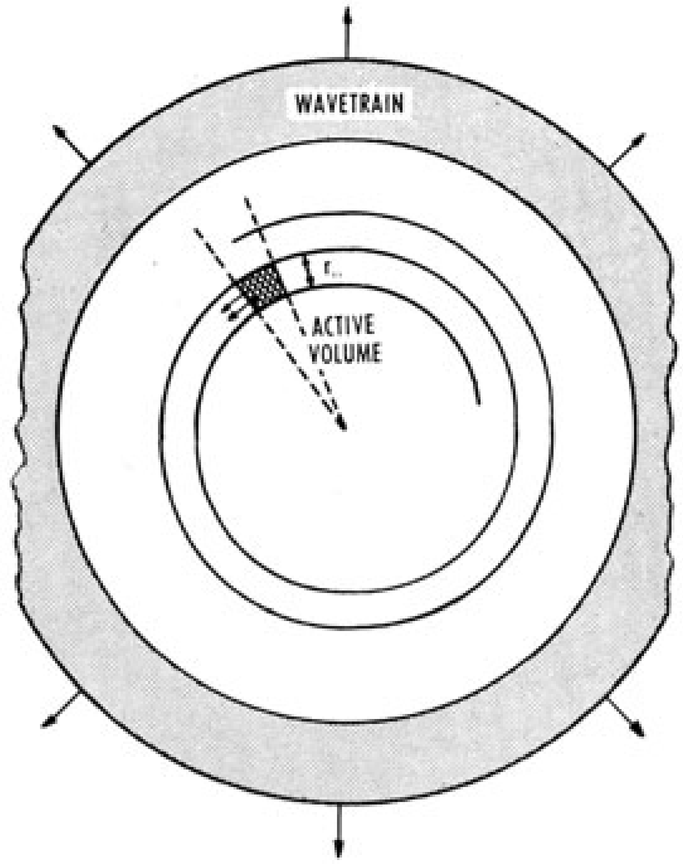
PULSE-TYPE SCANNING SONAR

Pulse-type scanning sonar equipment is, in effect, a combination of two types of ultrasonic echo-ranging and listening equipments operating simultaneously. One provides a continuous visual display of acoustic reception from all directions, and the other provides audio response from any desired single direction. The single-direction type is the exact equivalent of "searchlight" sonar. The function of detection by echo ranging is accomplished by transmitting a pulse of sound power in all directions and then scanning in azimuth for all echoes, which are made to appear as bright spots on a cathode-ray tube screen at the correct bearing and at a distance from the tube center proportional to the range. A more detailed investigation of a particular echo is obtained by training the audio system to the bearing indicated. The resulting audio output assists in identifying echoes, as well as in providing the signal for accurate range determinations. The reception of signals from noise sources, which is possible without transmission, produces a continuous radial pattern on the cathode-ray tube screen at the wave train and has half the extent of the wave train; its width, or extent in bearing, is limited by the directivity of the transducer on reception. Because the receiving beam pattern of the transducer is rotating, the active volume describes a spiral path. The radius of the spiral increases with half the velocity of sound; the speed of the active volume in the spiral path is much greater than this velocity.

In order for the active volume to encounter every possible target at some time, the beam pattern must not be rotated too slowly. Otherwise, the condition illustrated in figure 5-17 results; there is a dead area between the rings of the spiral traced out by the active volume. This dead area is shown unshaded, and echoes from targets in it are not received. In this case, the distance, S, of the spiral is greater than the ping length, r_o. If the beam pattern makes one revolution during the ping duration t_o of the signal, S equals r_o, and there will be no dead areas. If t_o is expressed in milliseconds and r_o in yards, r_o equals

proper bearing, and the audio character of this noise may be ascertained by training the audio system along that line or bearing.

At any instant the outgoing train of waves occupies a ring-shaped region (figure 5-16) marked "wave train." The radius of this region increases with time. Echoes can be returned to the transducer at a given instant from only a small region-the "active volume" shown as cross hatching in figure 5-16-which is determined by the ping length, r_o, and the angular width of the beam. This region is located at half the range of the



Wave train and active region for a rotating receiving beam pattern.

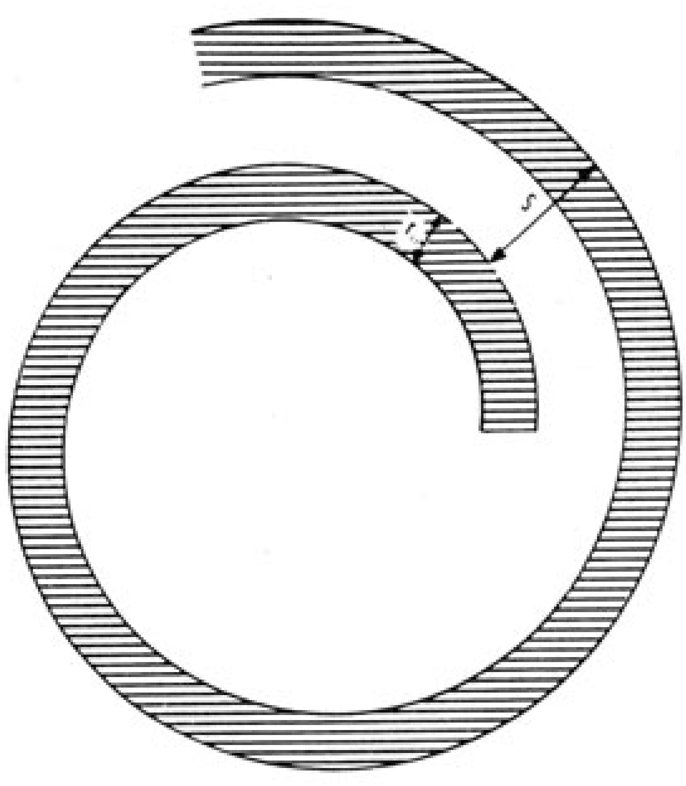


Figure 5-17. -Result of rotating a beam pattern too slowly.

ping length, provided the ping length is of the required order of magnitude.

The shortness of the echo duration is a consequence of the increased velocity with which the active volume moves. This increased velocity of the active region is the primary characteristic of pulsed scanning sonar.

The short duration of the echo, in its turn, has the following consequences:

1. Doppler discrimination is much impaired.
2. Because the spectrum of the short echo extends over many critical bandwidths of the ear, the advantage of the ear over other methods of perception is lost.
3. The pass band of the receiver must be at least wide enough to pass the short echo. This width involves increased noise levels.

$0.8t_o$, because t_o equals $2r_o/v$ seconds, where v is the velocity of sound.

Conversely, if the rotation of the beam pattern is fixed, the ping must have a duration of at least one revolution. Thus, if the beam pattern is rotated at 1,800 rpm, one revolution takes place in 33.3 milliseconds, and consequently the ping length must be greater than 0.8 times 33.3, or 26.7 yards. A value of 30 yards for r_o is safe if the active area is truly rectangular.

A consequence of the rotation of the beam pattern is that the echo will not have the same duration as the transmitted pulse. The echo will be received only while the beam pattern is passing the target. If the effective width of the beam pattern is θ degrees, the echo from a point target-that is, a target smaller than the active area-will be received during θ/360 of a revolution. For example, if θ equals 11° and the speed of rotation is 1,800 rpm, the duration of the echo is approximately 1 millisecond. Expressed in yards, the echo length, r_1, is 0.8 yard. The echo length, r_1, must be distinguished from the ping length, r_o. In every case, r_1 is smaller than r_o and is independent of the

4. The level of the reverberation, being determined by the volume of the active region, Is comparable to that of a standard sonar that transmits a ping of length r_o and is thus greater than for a ping of length r_1.
5. The coherence of the reverberation is comparable to that of sonars transmitting pings of length r_1.

These five effects tend to reduce the maximum range obtainable on a given target unless compensated either by a suitable device for detecting the echo or by the following effect.

6. The target strength of an extended object is determined by the size of the active volume and is therefore that which is characteristic of standard sonars transmitting pings of length r_o.

PLAN-POSITION INDICATORS

The high rate of rotation of the beam pattern makes it impossible for an operator to follow the changes in its heading with his unaided senses. This factor and effects 1 and 2 of the preceding list, make it necessary to use special devices to portray the echo and render the bearing and range of the target perceptible. These devices are called *plan-position indicators* (PPI).

The only device of this kind that is feasible for the high rates of rotation involved is a persistent-screen cathode-ray tube. The spot of this scope is made to describe a spiral path in synchronism with the active area. The path of the spot on the

screen is thus a map of the path of the active area. The brightness of the spot is controlled by the intensity of the received sound, so that an echo is seen as a brighter spot than the background of reverberation and noise. Because of the synchronization of the spot with the active area, the echo appears at the proper range and relative bearing on the screen.

If there are several targets in the field, they will be portrayed in their proper relative positions. Echoes obtained from reefs or sand banks appear on the screen as brightened areas. Thus a scanning sonar with a PPI presents the operator with a complete map of the underwater situation.

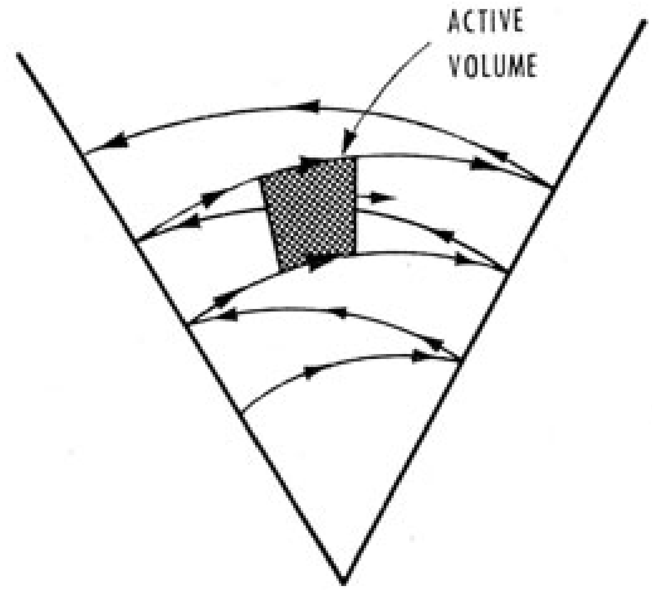


Figure 5-19. -Principle of scanning a sector rather than the complete horizon.

ROTATING-TYPE SCANNING SONAR

In theory, the transducer of a scanning sonar Could be directional and rotated about a vertical axis. However, the bulk of the transducer and the high rotational speeds required make this design impracticable.

Similar results can be accomplished by using a ring of stationary transducer units and connecting them in succession by means of a commutator (figure 5-18). In figure 5-18, however, only 12 transducer units are shown, whereas in practice 48 are used. Each of these is connected to one segment, *B*, of a stationary commutator. These segments are contacted by a rotating brush, *A*, which connects five or six transducer units to the receiver at any one time. As the brush rotates, these units are disconnected in succession and replaced by others farther along the ring. The result is that the receiving beam pattern of the array is markedly directional and rotates with the brush, *A*.

Because sliding contacts would generate too much electrical noise, a small gap is provided between the moving element, *A*, and the commutator segments, so that the brush is replaced by one plate of an electric capacitor. The received signal is thus connected to the receiver input by capacitive coupling rather than by conduction. This coupling, however, does not entirely eliminate commutator noise.

A second proposal for avoiding electric noise involves the elimination of all moving parts, and the use of electronic switches to perform the commutation.

In chapter 6 a typical scanning system is discussed in detail.

Sector-Scan Indicators

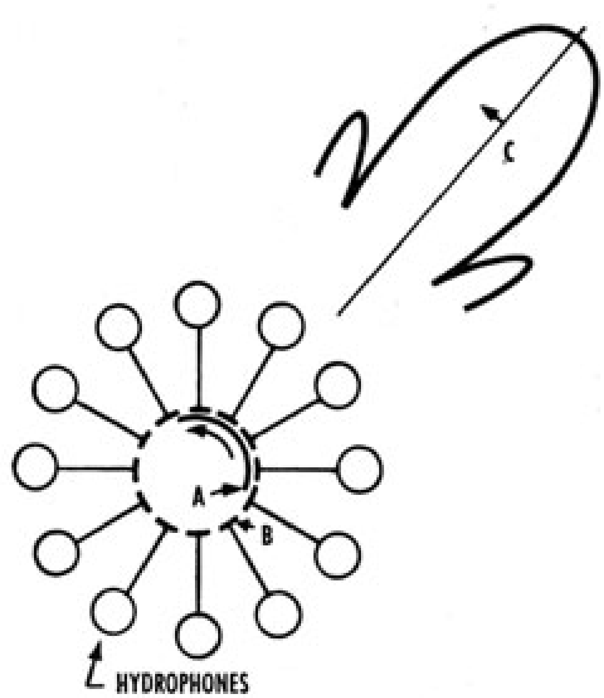


Figure 5-18. -Diagram illustrating cathode-ray scanning sonar.

The echo length resulting from the necessarily rapid motion of the active volume can be increased somewhat by scanning only a sector rather than the complete horizon. In this case the path of the active volume must be somewhat as shown in the schematic diagram in figure 5-19, and its speed can be reduced.

114

The oscillation of the receiving beam pattern can be accomplished by a modification of the principles already discussed in connection with the BDI.

F-M SCANNING SONAR

The long delay between the transmission of the signal and the reception of the echo, which is caused by the low velocity of sound, is a handicap in search operations. A pulsed scanning sonar utilizes this delay to scan all bearings, thus effectively increasing the speed of the active area.

The delay period also can be used to make other than single-frequency transmissions. Obviously, if the delay period is so used, a given echo must be associated with a given transmission. The idea can be illustrated very simply, as follows:

If the maximum practical range is about 3,000 yards, the maximum time delay is about 4 seconds. Suppose (1) that during these 4 seconds 8 pulses are transmitted at ½-second intervals, and target are stationary, this echo reproduces the constant frequency change of the transmitted signal. In figure 5-20, A, the solid curve is the frequency-time graph of the sawtooth signal; the dotted curve, that of the echo. Because of the time delay between transmission and echo, the sawtooth graph of the echo lags behind that of the signal by $2r/v$ second. The time interval indicated by T in figure 5-20, A, is the sawtooth interval. During a portion of the sawtooth interval, the echo frequency is less than the transmission frequency by a difference f'. During the remainder of the sawtooth interval the echo frequency is greater than the transmission frequency by a difference, f. This difference is illustrated by figure 5-20, B, in which the difference in frequencies between signal and echo is plotted as a function of time. The frequency difference, f, remains constant for relatively long periods and then jumps suddenly to the value, f'.

The frequencies are subtracted electrically by applying the heterodyne principle-a voltage tapped from the transmitter is combined with the echo signal in a heterodyne stage of the receiver. The

(2) that the frequency of each pulse differs from that of its predecessor by a stated amount. The frequencies of the pulses may form the tones of the major diatonic scale; if so, a musically inclined listener, on hearing an echo, can recognize its pitch and thus identify the ping responsible for it-provided, of course, that both source and target are stationary. Otherwise, doppler effect will alter the pitch of the echo. Some means has to be provided for recording the time and the transducer heading for each ping so that the range and bearing of the target can be determined.

Although this illustration is greatly oversimplified, it serves to introduce the discussion of a sonar that uses the f-m principle.

In practice it is simpler to change the frequency gradually rather than by abrupt steps. The frequency is decreased at a constant rate for some seconds; then, when it approaches the lower limit of the pass band of the receiver, it suddenly is increased to its original value, and the constant rate of decrease begins again. The principle is explained by the time graph of the transmitted frequency shown in figure 5-20, A. The transmitted sound frequency is a sawtooth signal. The intensity of the transmitted sound is kept constant during the transmission.

A target located in the sound beam returns a continuous echo and, if both the sonar and the range is determined from either of the

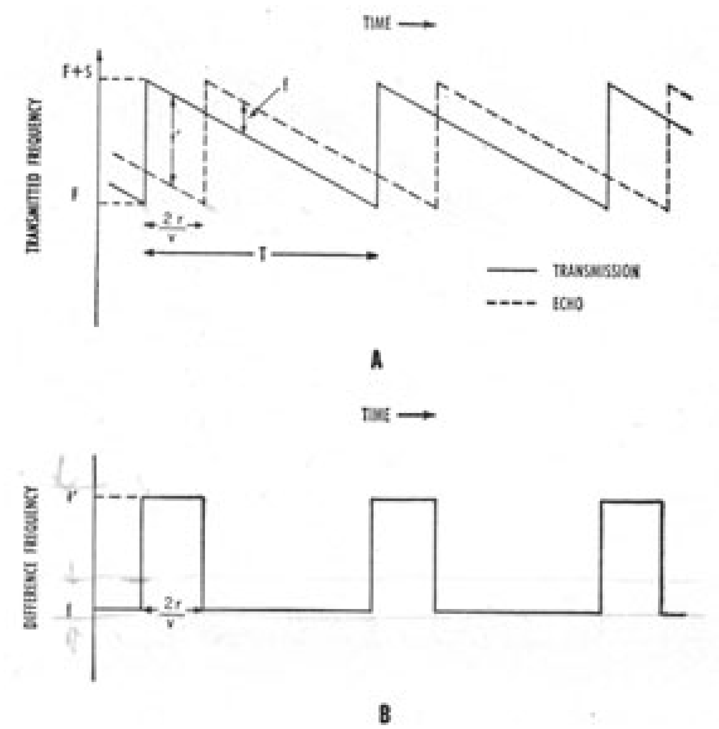


Figure 5-20. -Principle of f-m sonar. A, Transmitted frequency and the echo frequency as a function of time; B, frequency difference as a function of time.

frequency differences *f* or *f'*; the calculation of range from frequency difference is given later in this discussion.

Up to this point it has been assumed that there is only one target in the sound beam. If there is more than one returning echo, each echo will have its time graph of frequency, the displacement of which, relative to the graph of the signal, depends on the range of the target. The output of the receiver thus contains components of several frequencies-one pair of frequencies for each target in the sound field. This complex output must be analyzed into its components in order to determine the range of the several targets.

The active volume from which echoes are being received occupies the whole sound field. Furthermore, with omnidirectional projectors and hydrophones, no scanning of the field is possible. Thus, the basic principle of continuous transmission sonar in achieving effective detection is to employ a maximum size of the active volume, rather than to increase the speed of a small active volume as is done in pulsed-scanning sonar. The method just outlined for accomplishing this effective detection is called f-m sonar.

When used with a stationary projector and hydrophone, f-m sonar is not a bearing-scanning device. If it is used with an omnidirectional projector and a rotating receiving hydrophone, however, it becomes a scanning sonar with reduced active volume, but is different from the pulsed type.

Parameters of F-M Sonar

Relation between target range and echo frequency.-How the frequency difference between the echo and the transmitted signal determines the range is apparent from the following discussion and from figure 5-20.

The duration of one sawtooth waveform is T

2. The delay time for an echo from range, r, is $2r/v$ seconds.

3. In by $2r/v$ seconds the frequency therefore decreases by $(2r/v)(s/T)$ kc and the frequency difference, f, shown in figure 5-20, A, is

$f = 2r/v\ s/T$ kc. (5-14)

4. From equation (5-14), the range, r, is

$r = vT\ f/2s$. (5-15)

5. The frequency difference, f, is maintained for $T-2r/v$ seconds. At the end of this interval the transmitted signal has reached the bottom of the frequency sweep and returns to the top of the sweep. During a succeeding time interval equal to $2r/v$ seconds, the echo frequency is less than the transmitted signal frequency by f' kc (figure 5-20), where

$f' = s - f$. (5-16)

If the sawtooth interval, T, is several times greater than the delay time of echoes from the maximum range, frequency f is less than $s/2$ and frequency f' is greater than $s/2$.

6. The duration of frequency f' is considerably less than the duration of frequency f. Consequently, it is economical to ignore frequency f' and concentrate on the determination of frequency f.

Determination of frequency difference and range.-From equation (5-15) it is evident that f must be known to determine r. The range, r, is determined as follows:

Suppose the heterodyned output (the hydrophone output mixed with a sample of the signal) is passed through a band-pass filter that is centered at f kc, and that has a width w kc. This filter then passes an echo if its frequency lies within the band between f-

seconds. During this interval the frequency varies at a constant rate from $F+s$ to F; where s is called the *sweep* of the frequency. In one model, the QLA, F is 36 kc and s is 12 kc. T is usually from 1 second to 12 seconds.

The following relations exist between the several parameters:

1. The constant rate of frequency decrease is s/T kilocycles per second.

$w/2$ and $f+w/2$. Thus the sound energy admitted by this filter comes from a certain active area (figure 5-21), which is a sector of a circular ring.

A battery of such filters can be used to establish a series of channels, each of which is constantly alert to echoes from a certain active area. The dimensions of the area corresponding to a given filter can be calculated easily. The greatest range from which the particular filter under consideration will accept an echo is, from equation (5-15),

$$r_{max}=vT\,(f+½w)/2s,$$

and the smallest range is

$$r_{min}=vT\,(f-½w)/2s,$$

The radial extent, r_o, of the area is the difference between the two ranges, and thus

$$r_o=vTw/2s. \quad (5\text{-}17)$$

The other dimension of the active area can be determined by the range and the width of the receiving beam pattern; and from elementary geometry, its mean value is the product of the mean range and the angular width of the beam expressed in radians.

The dimensions of the active area are proportional to the sawtooth interval T and, in so far as they are seconds, s equals 12 kc, and w equals 35 cycles per second, then r_o equals 27.6 yards. Reducing the value T to 1 second would make r_o equal 2.3 yards, if v is taken equal to 4,742 feet per second.

As has been remarked, each of the channels is almost constantly alert to targets in the particular area associated with it. These areas are indicated in figure 5-21. As the active area of each channel is stationary, the whole sound field (figure 5-21) can be covered by making the areas of adjacent channels overlap slightly.

Because of the exclusion of frequency f', each channel normally is inert for a fraction of each sawtooth cycle. However, this fraction can be made as small as desired or even can be eliminated by means of a recent ingenious development.

The fact that the active areas are stationary may give the impression that the range cannot be determined as precisely as with pinging sonars. However, the precision is the same as for a ping length equal to r_o. The quantity r_o defined by equation (5-17) can be called the *effective ping length* of f-m sonar.

Range and bearing indication.-The range is read on an oscilloscope with a persistent screen. The filters corresponding to the various mean ranges of the

determined by T, are under the operator's control. For example, if T equals 12

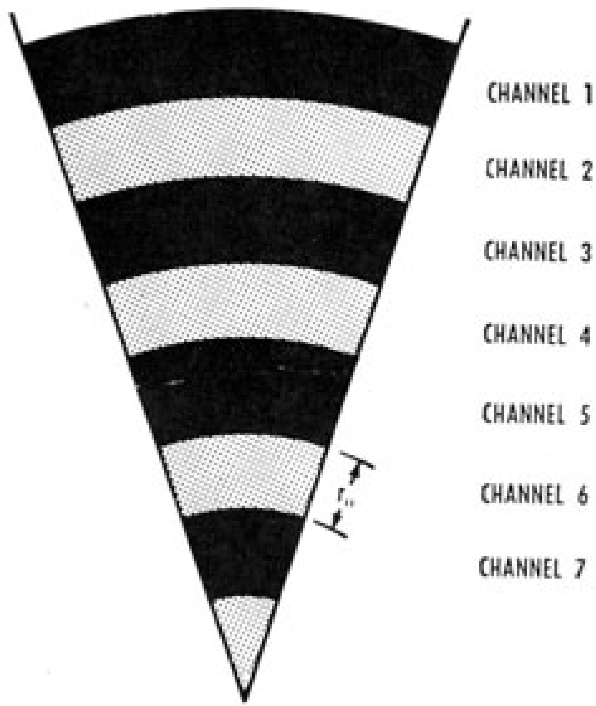


Figure 5-21. -Active areas associated with the individual channels in f-m sonar.

several channels are arranged so that their output brightens the trace of the cathode-ray tube at a point where the scalar distance from the center of the tube is proportional to frequency f and thus to the range. The bearing of the echo spot on the oscilloscope corresponds to the hydrophone heading.

Echo duration.-The duration of the echo depends on whether the hydrophone is stationary or is being rotated. If the projector is nondirectional, the echo received by a stationary hydrophone has a duration nearly equal to the sawtooth period, T. If the hydrophone is rotated, the echo duration is reduced and may become equal to the time required for the hydrophone beam to sweep across the target. The rate of rotation can be made as small as required to obtain an echo of any desired duration less than T. In this respect f-m sonar differs from the pulsed scanning sonars described previously.

The rate of rotation, however, cannot be increased beyond a certain critical value. This limitation is imposed by the use of filters, which require a finite time interval to respond fully to the

echo. The minimum time interval depends on the band-pass width, w, of the filter and must be greater than $1/N$ second, if w is in cycles per second. In other words, if the beam pattern of the hydrophone rests on a point target less than the time required for 1 cycle at the band-pass frequency, the echo will not be detected.

Suppose that the hydrophone is rotated at a rate of N rpm and that its beam width is θ°. A complete revolution requires $1/N$ minutes, which equals $60/N$ seconds. The beam occupies θ/260 of a revolution; thus the time required for it to sweep across a given point θ/360 times $60/N$, or $\theta/6N$ seconds. Hence, it is necessary that

$$\theta/6N > 1/w,$$

Call the corresponding ranges r_1, r_T, and r_E. In pulsed sonar the range indicated is always r_T regardless of the possible motion of sonar and target. The differences between r_1, r_T, and r_E are negligible. This fact can be verified quickly if it is remembered that a speed of 1 knot is equivalent to 0.56 yard per second, so that a speed of 25 knots involves an error of less than 50 yards in a range of about 3,000 yards.

None of the three ranges just defined is the range indicated by f-m sonar. Range r_i is defined by equation (5-15). In order to calculate r_i the following three ultrasonic frequencies must be distinguished:

from which it follows that N must be less than

$\theta w/6$.

For example, let θ equal 11° and w equal 35 cycles per second; then N must be less than 65 rpm. The echo duration for 65 rpm is 29 milliseconds; the corresponding echo length is 23 yards. If the rotation is slower the echo length increases.

Doppler range error.-It is clear that, because f-m sonar uses the frequency of the echo to determine the range of the target, the Doppler shift resulting from a possible relative motion of sonar and target introduces an error into the indicated range. The magnitude of this error must be evaluated.

F-m sonar is calibrated to give the indicated range, r_i, according to equation (5-15)-

$$r_i= vTf/2s.$$

This equation gives the correct range if the target is not moving, but it is necessary to calculate the error in r_i caused by the Doppler change of frequency when the range is opening or closing.

In all echo-ranging operations three instants of time must be considered. These instants are (1) t_1, the time at which the primary sound was transmitted from the transducer; (2) t_T, the time at which the echo was reflected from the target; and (3) t_E, the time at which the echo was received.

If there is any relative motion of sonar and target, the range is different at these three times.

1. F_1, the frequency that was being transmitted at time t_1.
2. F_E, the frequency that was being transmitted at time t_E.
3. F'_E, the frequency of the echo that was being received at time t_E.

The quantity, f, of equation (5-15) is obviously F'_E-F_E hence,

$$r_i=vT/2s(F'_E-F_E). \quad (5\text{-}18)$$

The frequencies F_E and F'_E must be examined more closely. To simplify the calculations, the sonar is assumed to be stationary. No error is introduced by this assumption if dR of equation (5-20) is interpreted as the range rate. When the target reflects the sound its range is r_T, and the transmitted frequency, by the time t_E, has been reduced by $(2r_T/v)(s/T)$: The possible motion of the target will not affect this quantity. Thus, the frequency being transmitted at the instant when the echo is received is

$$F_E=F_1-(2s/v_T)r_T \quad (5\text{-}19)$$

The frequency of the echo, on the other hand, is affected by the motion of the target. From the theory of the Doppler effect, the value F'_E is approximately

$$F'_E=F_1\pm(2dR/v)F_1 \quad (5\text{-}20)$$

where dr is the range rate of the target. If equation (5-19) is subtracted from equation (5-20),

$$F'_E-F_E=(2s/vT)r_T\pm(2dR/v)F_1, \quad (5\text{-}21)$$

and if equation (5-21) is substituted into equation (5-18),

$r_i = r_T \pm (dRT/s)F_1$. (5-22)

The error in the indicated range, as shown by the last term in equation (5-22), is therefore proportional to the velocity of the target and is zero only for stationary targets.

For example, let T equal 12 seconds; *s* equal 12 kc, and *F*, equal 36 to 48 kc; then

TF_1/s=36 to 48 seconds.

The range error is thus the distance moved by the target in 36 to 48 seconds. The larger error occurs when the sawtooth frequency, F_1 is high and the smaller error occurs when it is low. The distance traversed in 48 seconds by a submarine at 10 knots is slightly more than 200 yards.

Note that the error is also proportional to the sawtooth period. Thus, if *T* had been 1 second in the example, the range error would be the distance moved by the target in 3 to 4 seconds, or about 20 yards, at a speed of 10 knots.

This range *error* is very similar to the range *correction* which must be made in determining the time to fire on a moving target. It has been proposed to utilize this similarity so that the indicated range of f-m sonar can be used without this correction in fire control problems. For this application, it is essential for the frequency to increase rather than to decrease during each sawtooth period.

Location of Small Objects

The echo-ranging equipment in use at the beginning of World War II was designed for the detection of relatively large submarines. As the war progressed it became imperative to design equipment for the detection of mines and other small objects. The standard test object in this development work was a sphere 3 feet in diameter. The target echo strength of such a sphere is some 20 db lower than that of a large submarine. Because of this small echo strength, the ranges in small-object detection generally are comparatively short and thus are limited by reverberation rather than by background noise.

In order for an echo to be detected against a background of reverberation, the total echo strength of all the scatterers in the active volume of a ping must be less than the echo strength of the sphere; otherwise, the reverberation intensity is greater than that of the echo and masking prevents available parameter for reducing the size of the active volume.

The use of short pings is thus a characteristic of many sonars designed for small-object detection. In this phase of echo ranging, the reasons for the success of scanning sonars-which do not use short pings-are not clearly understood; however, their success probably depends on the plan-position presentation of the echo, or upon the limitation of the active volume by the scanning process.

RATIO OF ECHO TO REVERBERATION AS A FUNCTION OF PING LENGTH

According to the theories that have been previously explained, the reverberation intensity is proportional to the ping length, r_o. The echo level, on the contrary, is independent of ping length except when

detection.

The echo strength of the reverberation can be decreased by reducing the size of the active volume. There are two ways in which this decrease can be accomplished: (1) The ping length can be decreased; and (2) the beam can be made narrower. The second method is not suitable for shipboard installations because it involves a decrease in the effective search area and thus causes great difficulties in maintaining contact with the target. This situation leaves only the ping length as an the ping length becomes less than the variation in range for different parts of the target. If the target is complicated its target strength is less for short pings. This condition exists because the echoes from some parts of the target no longer overlap those from other parts. However, if the target has a smooth surface, with no irregularities of dimensions comparable to 1 wavelength of the sound, this reduction in target strength will probably not occur. The theory of echo formation has not been worked out sufficiently to cover this point.

The results show that the echo-reverberation ratio increases with decreasing ping length even when the ping length is as small as one-eighth the diameter of the sphere. They also show that this ratio generally decreases with increasing range out to 400 yards, thus supporting the idea that reverberation, rather than background noise, is the limiting factor in this work. Such results are rather surprising because a wide-band receiver is necessary for the use of these very short pings. The qualitative distinction between reverberation and noise largely disappears at these ping lengths, for both reverberation and noise sound alike to the listener and have a similar appearance on an oscillogram. Consequently, these experiments are the best evidence that reverberation, and not noise. is the masking agent. This high level of reverberation is due to a combination of factors, principally the shallow water and the short range. Both are typical of the conditions under which the equipment must operate.

Variation of Gain

The optimum gain setting is the one which makes the masking background just audible. If the gain is less than the optimum, weak signals will not be heard even though they are stronger than the background. If the gain is much greater than the optimum, there is danger that a signal will overload the amplifier, resulting in (1) distortion and (2) a reduction of the signal-background ratio in the airborne output.

This situation is complicated when reverberation is the masking background, because the reverberation level varies greatly during the period following transmission.

are common in radio receivers. Such circuits are called *automatic volume controls* (AVC). The use of AVC circuits in sonar receivers, however, has proved to be very disadvantageous. AVC circuits can be adjusted to respond rapidly or slowly to changes in the input. If they respond to the rapid fluctuation of reverberation, they also respond to the change in intensity due to the echo. This response is unavoidable, because the reverberation signals last about as long as those of an echo from a point source. With this adjustment the AVC reduces the gain during the time the echo is being received-an obviously undesirable situation. If, on the other hand, the AVC is adjusted so that it does not reduce the gain during the echo. it becomes so sluggish that it fails to respond to the slower changes in mean

The obvious solution for this problem is to devise a sonar receiver in which the gain continuously increases during the period following transmission of the ping. The receiving circuits for accomplishing this *time variation of gain* (TVG) are controlled by the discharge of a capacitor that is charged (During the transmission. The rate at which the gain increases can be controlled by altering the resistance of the discharge circuit. The total increase in gain can be adjusted by altering the voltage to which the capacitor is charged.

Although TVG improves the operation of the echo-ranging equipment, it fails to meet all the requirements. One disadvantage of TVG is that the gain is increased in a regular manner. This regular increase of gain would be satisfactory if reverberation decreased in an equally regular manner, but reverberation decrease is not always regular, especially in shallow water. Consequently, the possibility of using the background to control the instantaneous gain was explored.

Circuits that can control the gain automatically reverberation level-for example, to the peak of bottom reverberation.

A compromise solution, called the *reverberation-control of gain* (RCG), has been developed. RCG is similar to TVG in that the gain constantly increases during the period following transmission. It is similar to AVC in that the momentary level of the receiver input controls its operation. In RCG, however, it is the *rate of increase of gain* that is controlled and not the *gain* itself. It is obvious that an RCG circuit cannot reduce the gain at the instant the echo arrives; it merely reduces the rate at which the gain increases while the echo is being received. Thus, it does not have the disadvantage of an AVC circuit.. RCG responds somewhat, to the special characteristics of reverberation at a specific time and place and thus does not have the disadvantage of a TVG circuit that is improperly adjusted for the momentary conditions.

Maintenance of Close Contact

The development of deep-diving, high-speed submarines presented a serious problem. The echo-ranging equipments in use at, the beginning of World War II were not capable of maintaining sonar contact on a deep submarine when the range was closed. The deeper the submarine was operating, the greater the "lost-contact" range. Sometimes the contact was lost at 600 yards because the submarine passed under the lower limit of the sound beam. A loss of contact made it impossible to attack successfully, because the high speed of the submarine enabled it to be a considerable distance from the point of lost contact.

transducer is depressed toward the deep target, and hence away from the surface, the echo level can be increased and the surface reverberation decreased. A sonar equipped with the tilting-beam transducer can also determine the depth of a target, as will be explained later.

Another method, known as *maintenance of close contact* (MCC), changes the connections of the transducer elements so that the beam becomes very broad in the vertical plane. This change is accomplished by a switch so that at long ranges the beam pattern is undisturbed because the distortion of the beam reduces its efficiency. This inefficiency is inconsequential at the shorter ranges, where contact would be lost if the MCC feature were not used.

This problem can be overcome by mounting the transducer like a searchlight, so that it can be (1) rotated about, a vertical axis and (2) tilted about a horizontal axis. If the axis of a tilting beam

Depth Determination

Both the depth and the horizontal range of the target can be determined by mounting a transducer so that the beam can be tilted in the vertical plane. The geometry of the situation is shown in figure 5-22. The range indicator of the sonar

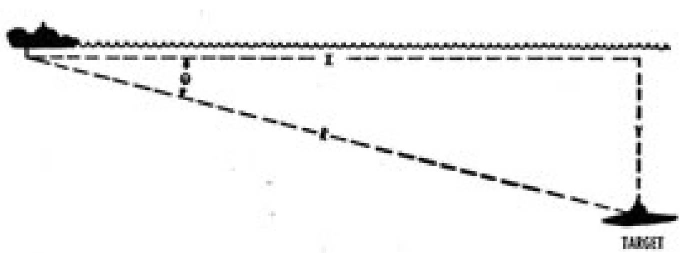


Figure 5-22.-Geometry of depth determination.

equipment shows the *slant range*, R. The *depth* of the target below the projector is Y and its *horizontal range* is X. If the angle of tilt, θ, is known, the values of X and Y can be calculated from the equations

$X = R \cos \theta$
and
$Y = R \sin \theta$. (5-23)

Various automatic or semiautomatic methods of performing this calculation have been devised.

REFRACTION ERROR

Equations (5-23) assume that the sound rays are straight lines, as shown in figure 5-22. If the rays are refracted (figure 5-23), the values computed from this equation are X_o and Y_o instead of the actual values X and Y. The errors $Y - Y_o$, and $X - X_o$ can be quite large, especially when there is a marked downward refraction.

The errors arise from two causes: (1) The sound does not travel in a constant direction, and (2) it does not travel at a constant speed. The determination of the corrections to be applied is similar to a problem in exterior ballistics. The problem can be solved by the same methods, but when there is a marked thermocline the magnitude of the correction required is increased, as is also the accuracy required in making the approximate calculation. Semiautomatic methods have been developed to speed the application of the correction during combat.

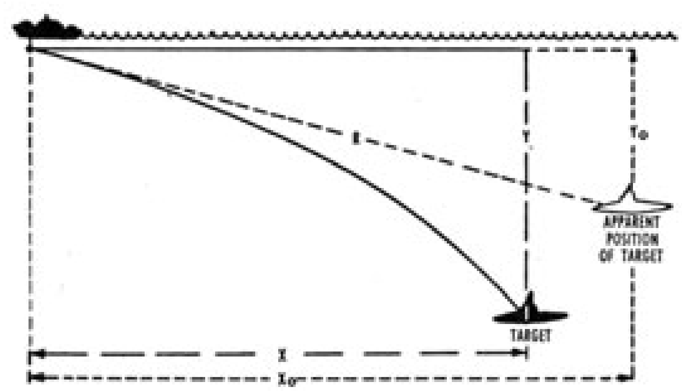


Figure 5-23. -Effect of downward refraction on depth determination.

Integrated Sonar System

The integrated sonar system is an attack system. It is composed of several sonar and fire control equipments operating in a reciprocal information and control network.

The antisubmarine system used at the outbreak of World War II consisted of a single echo-ranging equipment. From the equipment, the *bearing* and *slant range* of the target could be obtained. The procedure of "crossing the target," previously explained, established the *bearing rate*. If this information was plotted on a maneuvering board and if a stop watch was used to time the development of the attack, the *range rate* could be determined, as well as the approximate course and speed of the submarine. The conning officer guessed the depth of the submarine from the range at which sonar contact was lost because of the targets passing under the sound beam.

By the time the *lost-contact range* was reached, the ASW vessel was on its attack course and was proceeding to saturate the area with depth charges. This method was fairly effective against the old type of submarine with riveted construction, because the pressure hull could be ruptured by a nearby explosion. One disadvantage of delivering a depth-charge attack against a modern submarine is that the pressure hull with its all-welded heavy-metal construction can withstand anything but a direct hit or a very near miss.

Other disadvantages of delivering a depth-charge attack on a modern submarine are as follows:

1. Because the charges sink very slowly, a large lead must be taken to place them well ahead of the target. Thus, contact is lost before the charges are dropped. The time between lost contact and the point of dropping the charges, plus the time required for them to sink to the proper depth, gives the submarine time to evade.

2. After contact is lost on a high-speed target, it is difficult to regain. When a depth charge explodes, it sets up a turbulent area that returns strong echoes. The submarine may escape from the area because echoes from the turbulent area mask the echoes from the submarine.

To attack a modern submarine successfully the following requirements must be met:

1. After contact has been established it must be maintained until the submarine is put out of action.
2. Only ahead-thrown weapons that explode on contact should be used.
3. The depth and horizontal range of the target must be determined as well as its bearing, speed, and course.

An integrated system has been designed to meet and to coordinate these requirements. This system requires two echo-ranging equipments-one equipment for azimuth search, which furnishes the *slant range* and *bearing*, and the other for depth search, which furnishes the *depression angle* of the target. From the slant range and depression angle a sonar resolver computes the depth and horizontal range. To obtain the most accurate sonar information possible, the equipment must be stabilized so as to remove the components of own ship roll and pitch. Such a system requires a stable element and a stabilization computer. An underwater battery fire-control system is used to solve the fire control problems of the (1) *target course* and speed, (2) course that own ship should steer, and (3) time to fire. Included in the system are plotting devices for keeping track of the attack as it develops.

122

[Previous Chapter] [Sonar Home Page] [Next Chapter]

CHAPTER 6

SURFACE-SHIP ECHO-RANGING EQUIPMENT

Constant effort and research are devoted to the development and improvement of naval sonar and associated equipment. As more efficient equipment is perfected old equipment is replaced as rapidly as possible consistent with production and money available. There is an unavoidable time lag between the development and installation of new equipment in ships of the fleet and inclusion of such equipment in a publication such as this. At the time you use this book, some equipment discussed in this and succeeding chapters may not be the equipment currently being used in the fleet. However in most cases circuits and operating principles will be similar to those found in current equipment. Equipment manuals for specific equipment should be studied thoroughly by personnel responsible for repair maintenance and operation of that equipment.

MODEL QGB

The model QGB sonar equipment, shown in figure 6-1, is typical of modern searchlight equipment. This type of equipment is designed for installation aboard destroyer escorts and destroyers. The QGB consists of a transmitter, a receiver, an indicating range recorder, a bearing deviation indicator, and a transducer with an associated hoist-train mechanism.

The system operates at a frequency determined by the resonant frequency of the magnetostriction transducer. Available frequencies are 20, 22, 23, and 26 kc. The transmitter and receiver both cover the frequency range of from 17 to 26 kc. To change the operating frequency of the system, and still have it operate efficiently, the transducer must be changed. Because magnetostriction training control, and the keying control unit are contained in a console, which is the sonarman's operating station. The indicator panel of the console is shown in figure 6-2. This console is housed in the sonar control room. The transmitter and its power supply are contained in a cabinet in the sonar equipment room. The hoist-train mechanism is located in the lower sound room.

Receivers and transmitters will be discussed in chapters 7 and 8, respectively.

The transducer, with its training and hoisting mechanism, generally is installed so that the unit is parallel to the fore-and-aft axis of the ship. The transducer, which can be rotated through a maximum of about 840°-2 1/3 revolutions-operates over the entire port and starboard sectors of the vessel.

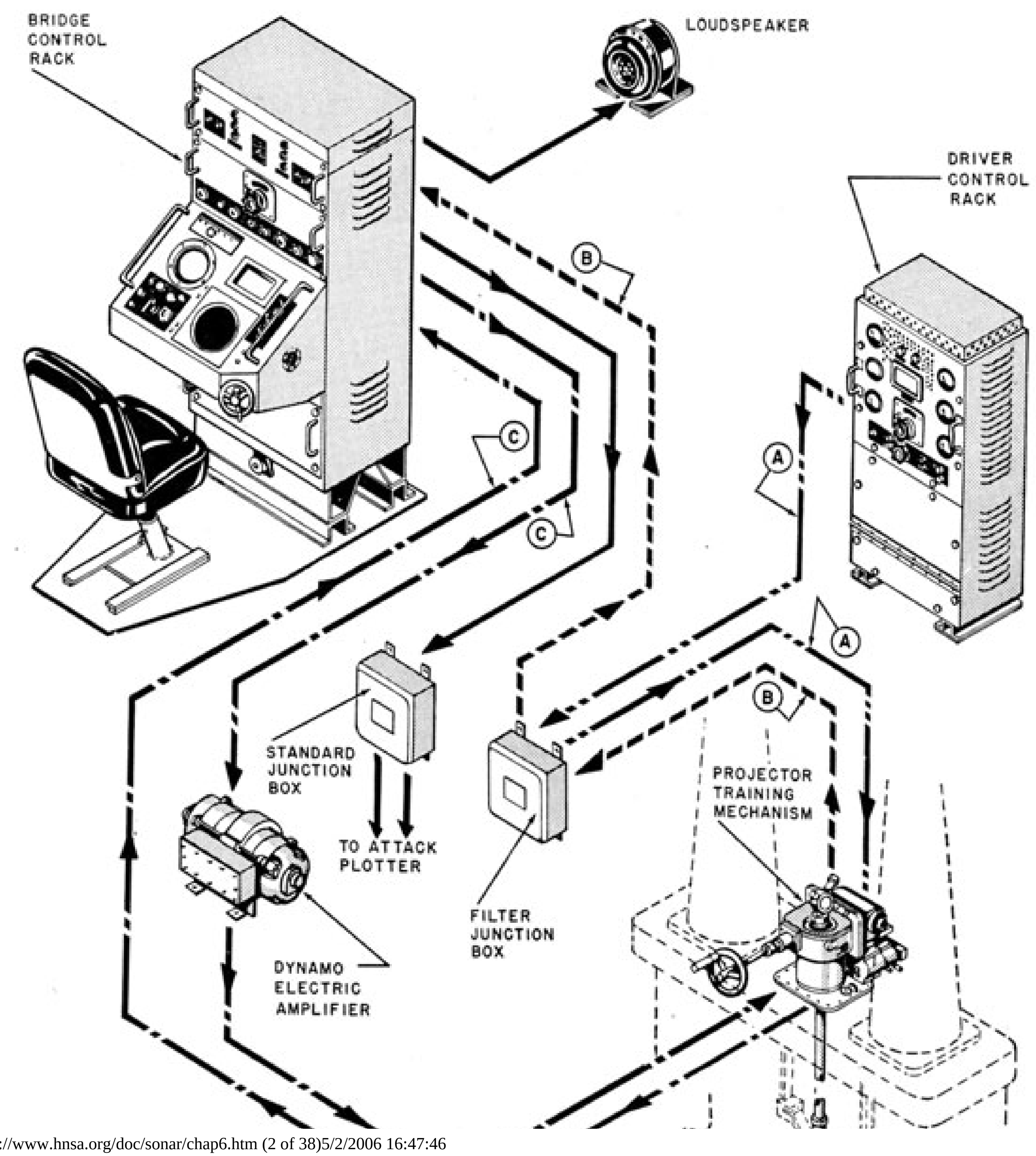
The transducer itself is mounted in a *sea chest* and the transducer and its hoisting units are raised and lowered inside this chest. When in the raised position, the sound dome seals off the sea chest so that water cannot enter the ship if the top of the chest were removed. Sealing off the chest in this manner permits installation, removal and servicing of the transducer and its training mechanism while the ship is at sea or in port. All other units of the system are accessible from the interior of the ship.

The bearing scale on the transducer shaft in the lower sound room is adjusted to read zero when the transducer is directed dead ahead. The bearing is remotely repeated at the operating position in the sonar control room by a synchro repeater system. The repeater on the operating console indicates either true or relative bearings.

transducers have sharply resonant characteristics, the system must be adjusted to the resonant frequency of the transducer.

The receiver, bearing-deviation indicator, indicating range recorder, range indicator, remote

The transducer is trained by rotating a handwheel on the console, geared to a control transformer. The output of this control transformer is the error signal for the training control amplifier that controls the amplidyne generator.



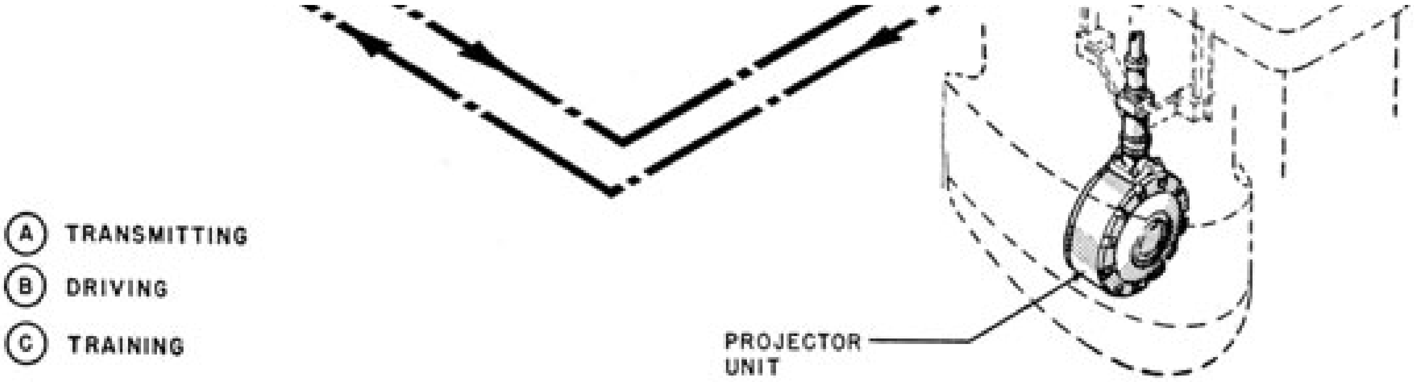


Figure 6-1. -Pictorial diagram of the QGB system.

124

After the *maintenance-of-true-bearing* (MTB) feature is switched *on* any change in the ship's heading causes the transducer to rotate by the same amount but in the opposite direction. Thus, the transducer remains on the same true bearing regardless of changes in the ship's heading.

Keying of the equipment can be controlled by any one of the four means-(1) the range recorder located in the console of the equipment, (2) an external tactical range recorder, (3) a hand key on the console, or (4) a multivibrator that is a part of the keying unit.

The keying intervals are arranged and controlled as follows. When keying is controlled by the sound-range recorder, two scales are available-one at 1,500 yards and the other at 3,750 yards. To conserve recorder paper when there is no target, a multivibrator takes control. The multi-vibrator has two time rates-one at a scale of 3,000 yards and the other at a scale of 5,000 yards.

During search operations one of the last two scales is selected depending upon sound conditions.

The keying interval and the conditions of manual keying or listening are selected by a rotary switch on the console.

The cycle of operation of the QGB is as follows: (1) A keying signal is delivered to the keying unit which energizes the keying relay. (2) The relay transfers the transducer from the receiver input to the transmitter output. (3) The transmitter impresses an r-f voltage of the correct frequency across the transducer at the proper energy level. (4) The transducer converts this energy into sound power, and emits it in a narrow beam along the bearing to which the transducer is trained. (5) Immediately after the keying period the relay restores the transducer connections to the receiver input. (6) Any sound energy that is returned to the transducer is converted into electric energy and applied to the receiver input.

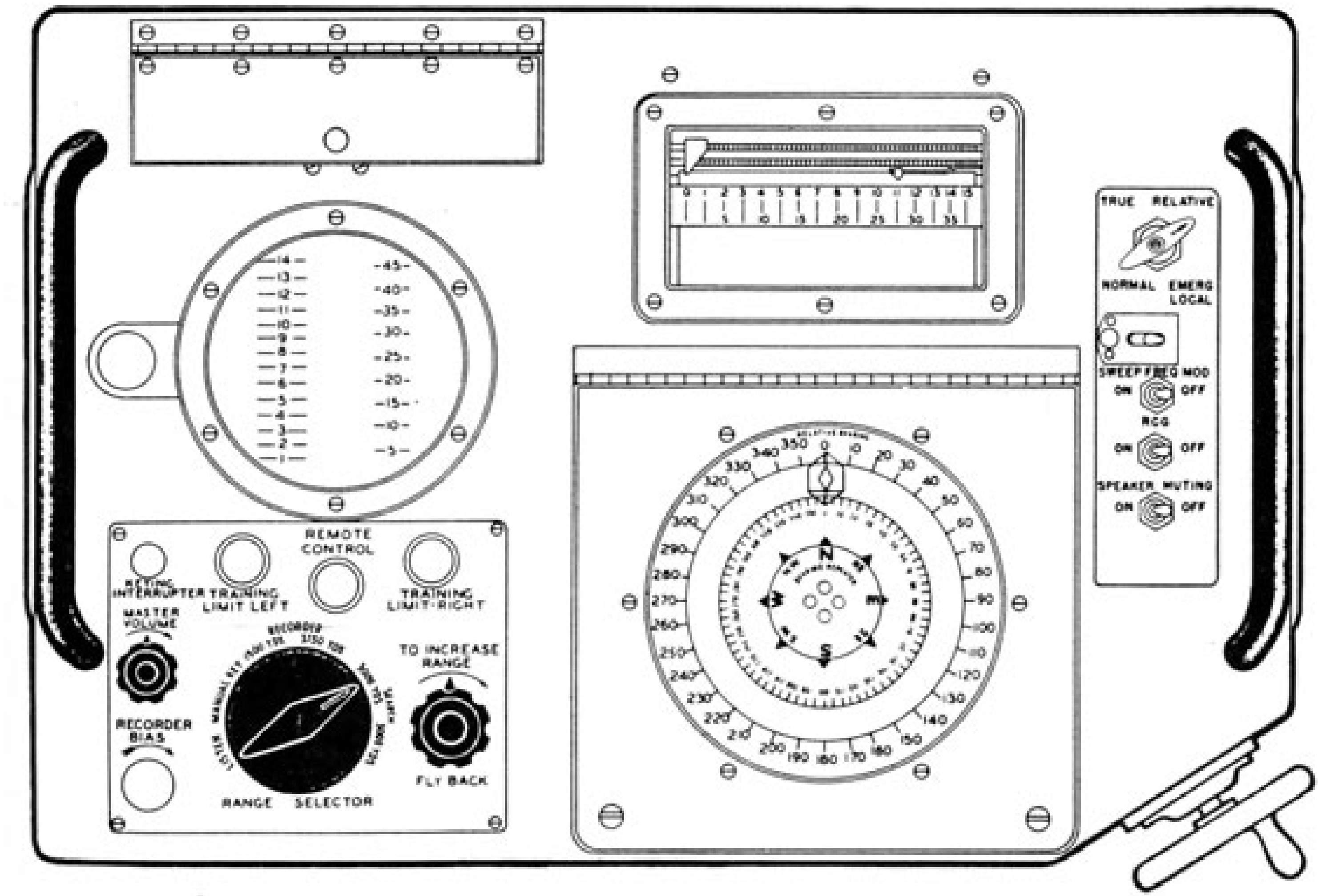


Figure 6-2. -Indicator console of the QGB system.

239276°-53-9

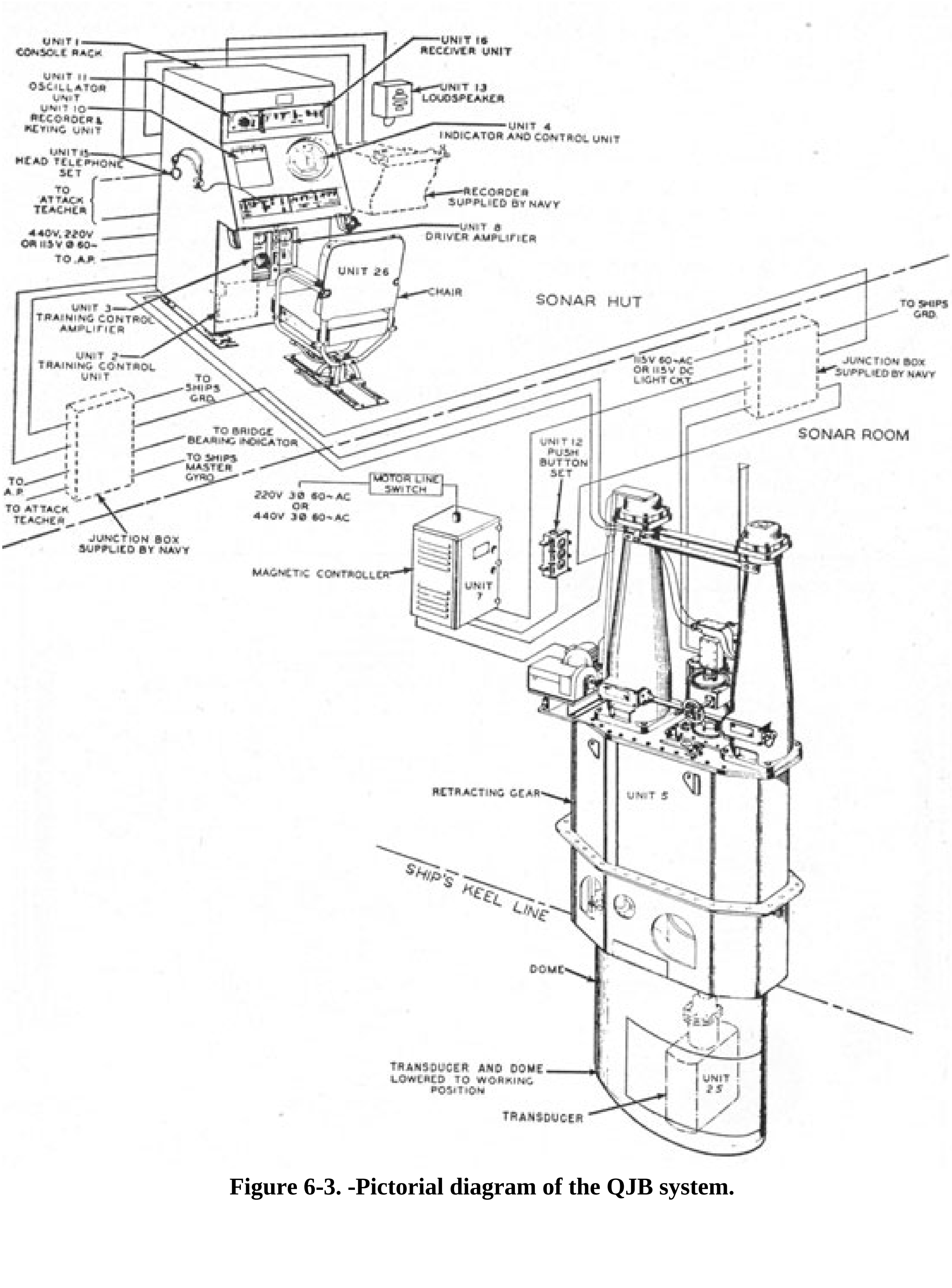


Figure 6-3. -Pictorial diagram of the QJB system.

In the QGB, the receiver output is supplied to a loudspeaker or headphones to allow the operator to identify the signals. This signal voltage may also be used to mark the paper in the recorder and make a permanent record on the chart. A third use of the signal is to deflect the beam of the bearing deviation indicator (BDI).

MODEL QJB

The model QJB, shown in figure 6-3, is a searchlight sonar equipment also intended for installation in destroyers and destroyer escort vessels. This equipment is smaller than the QGB, and all its electronic units are housed in a single console. The QJB transmitter is much smaller than the QGB transmitter, because of the lower power requirements of the QJB. Although the transmitted power is less, the resultant echoes are equal to those of the QGB due to the higher sensitivity of the crystal transducer of the QJB. The transmitter is mounted in the lower right section of the console. The other units in this console are the receiver, keying unit, bearing deviation indicator, indicating range recorder, remote training control, and power supplies.

The receiver is of the sum and difference type and has both time-varied-gain (TVG) and reverberation-controlled-gain (RCG) features.

The QJB utilizes the unicontrol-oscillator system for tuning the receiver

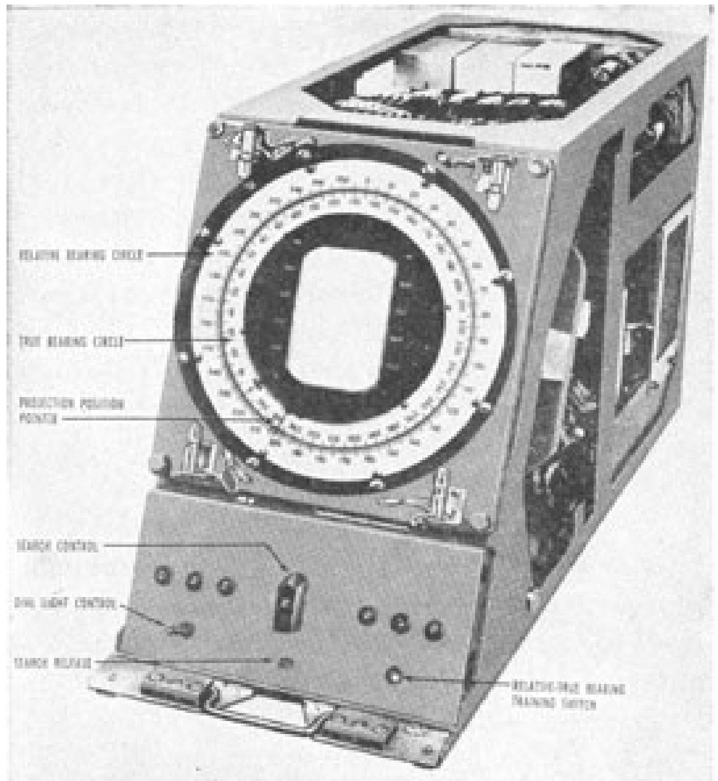


Figure 6-4. -BDI and remote bearing indicator of the QJB console.

The transducer is made up of ADP crystals mounted on a steel plate. These crystals project perpendicularly from it a distance equal to one-quarter wavelength of sound in the crystal medium. Opposite the crystals on the reverse side of the mounting plate are steel rods extending a distance equal to one-quarter wavelength of sound in the steel medium. These steel rods are longer than the crystal units because the velocity of sound is greater in steel than in crystal. This system of mounting the crystals with the resonating rods results in greater power output than if the crystals were used alone. It has the disadvantage of making the transducer frequency sensitive.

This combination forms a half-wave system rigidly mounted in the center. In this type of system the mounting plate is stationary, and the crystals vibrate in a direction perpendicular to the plane of the mounting plate. The crystals are connected so that they all vibrate in time phase-all crystal surfaces move in the same direct ion at the same time. The results of this arrangement approach the theoretical results of the piston type

and transmitter. The operating frequency need not be tuned carefully to the frequency of the crystal transducer because the transducer has a rather flat resonance curve-about 6 kc wide.

The chief difference in the appearance of the QJB console as compared with the QGB is the manner in which the BDI cathode-ray tube is mounted. The QJB mounting is shown in figure 6-4. The tube is located in the center of the bearing circle so that the operator does not have to shift his eyes away from the bezel indicating the transducer bearing when he wishes to see the BDI indications. To mount the BDI cathode-ray tube in the center of the bearing indicator-which consists of a true-bearing circle and a bezel ring that indicates the transducer bearing-a complicated mechanical system is required.

The keying-control unit and indicating range recorder are conventional.

of vibrating source.

The crystal array is housed inside a rubber sound window. The space inside the sound window that is not occupied by crystals is filled with castor oil that has been treated to remove all air and moisture. This castor oil protects the crystals from damage by moisture, because it excludes water.

The only other unit of this equipment that is not located in the console is the retracting gear, which is in the hull of the ship near the keel. This retracting gear is similar to that of the QGB

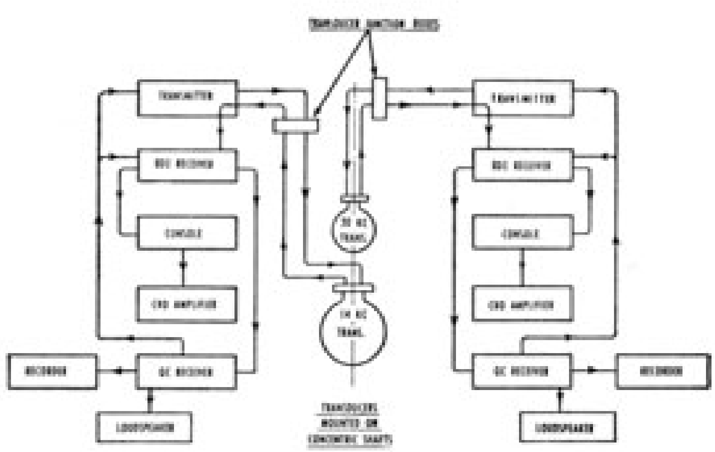


Figure 6-5. -Block diagram of the QGA system.

external tactical recorder can be used to control the transmission of either equipment or both of them.

equipment. However, it is slightly smaller because the QJB transducer is smaller.

MODEL QGA

Model QGA (figure 6-5) is another searchlight sonar system designed for installation on destroyers. The system has two complete sonar equipments that are practically identical. One operates on a frequency of 14 kc; the other, on a frequency of 30 kc. The 30-kc transducer can be tilted downward from an angle of 0° to an angle of 45° below the deck. This feature is of value when the sonar vessel is approaching a deep target.

The QGA consoles are similar to the QGB console. They are installed side by side in the sonar control room. The two equipments of the QGA are capable of independent operation, or they may be slaved by a control on the 14-kc console. An

The receiving system for each console consists of an audio receiver and a BDI receiver. The transmitters are conventional r-f amplifiers. A unicontrol-oscillator system tunes the receivers and transmitter of each unit.

The magnetostriction transducers are mounted on concentric shafts that are hoisted and lowered together. The 30-kc transducer is smaller because of its higher frequency. It is mounted over the 14-kc transducer. The training mechanisms are arranged so that the transducers can be trained independently of each other.

Scanning Sonar Equipment

MODEL QHB-a

The model QHB-a scanning sonar equipment (figure 6-6) is an ultrasonic, magnetostrictive, echo-ranging-listening equipment that provides a video presentation of acoustic reception from all directions and an audio presentation of reception on any selected bearing.

In the echo-ranging condition, the QHB-a transmits a pulse of sound power in all directions, and then scans or samples all echoes so as to produce on the screens

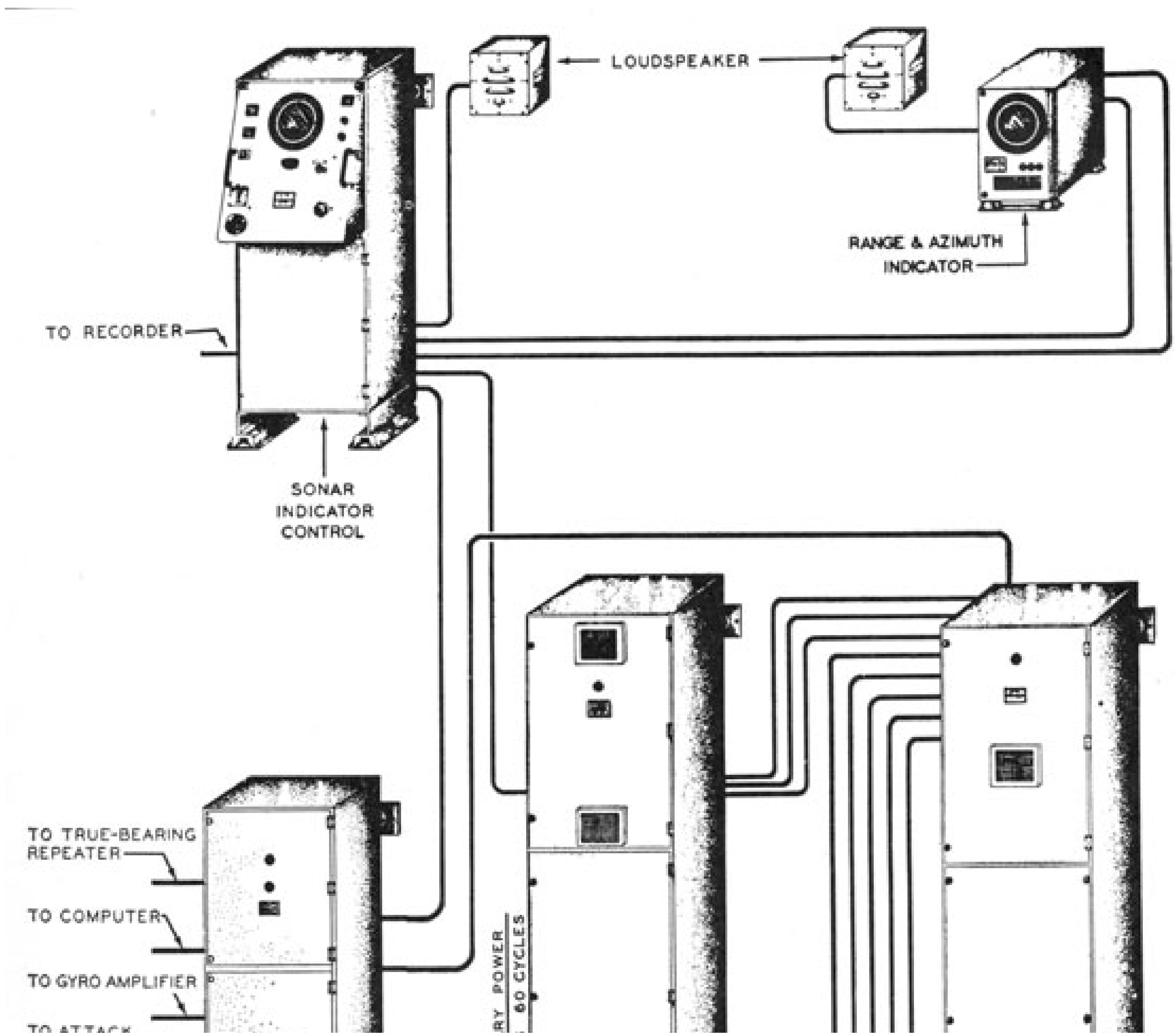
still alert in all directions. It scans 26-kc ultrasonic frequency noise and produces radial patterns on the screens of the cathode-ray tubes, from which the true bearing of any noise source can be obtained. Simultaneously, the audio-channel sensitivity pattern may be trained on any noise source for determination of its character.

System Line-Up

The system employs a single transducer for both transmission and reception. It contains 48 electrically independent hydrophones, which are arranged symmetrically along the periphery of a nontrainable cylinder. During the transmission pulse of 35 milliseconds these hydrophones are connected in

of associated cathode-ray tubes a plan-position indication of all echoes received. Simultaneously, the audio-channel sensitivity pattern may be trained in any desired direction for aural recognition of the characteristics of any of the echoes, as well as for determination of the range by means of a range recorder.

In the listening condition, automatic transmission is omitted. However, the video channel is parallel by the receive-transmit switching relay so that the acoustic power is transmitted simultaneously in all azimuth directions. Immediately after the transmission pulse



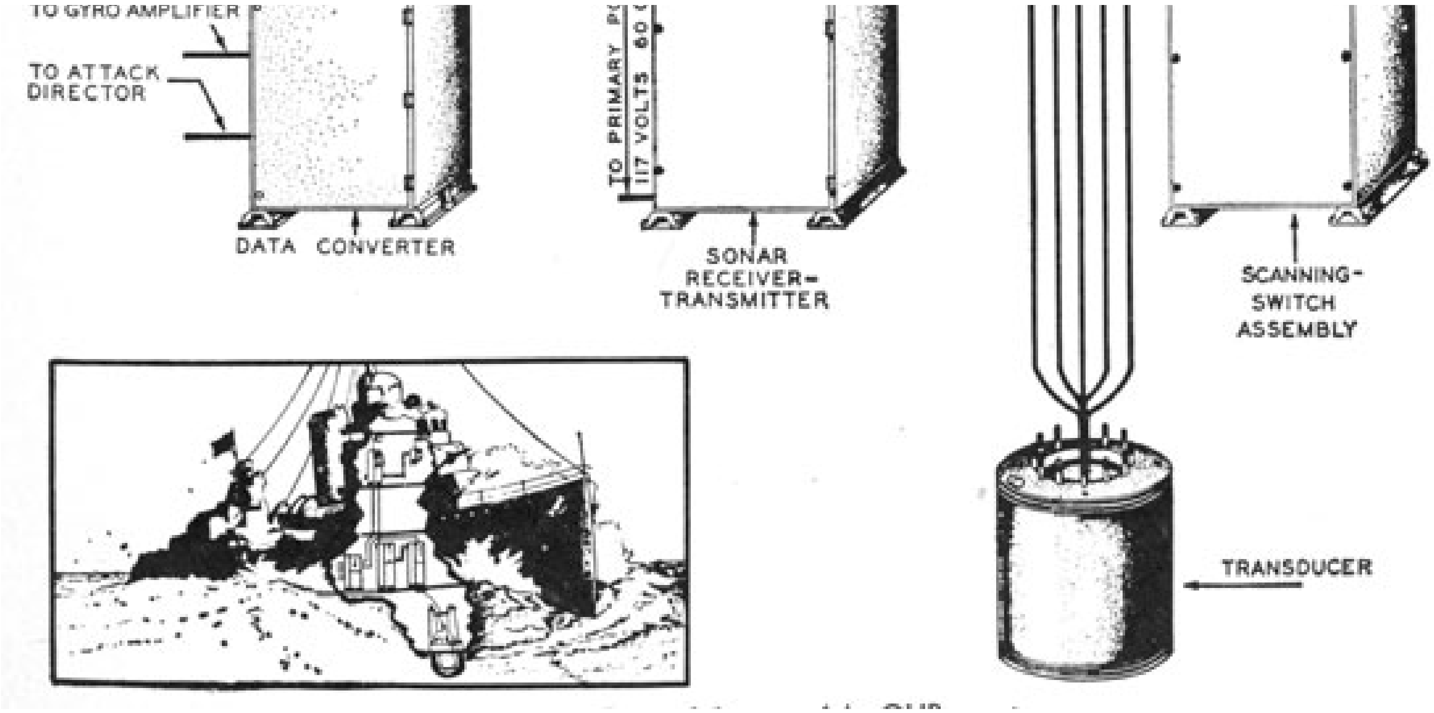


Figure 6-6. -Pictorial diagram of the QHB-a system.

the receive-transmit switching relay returns the circuit to normal so that the hydrophones are again independent. Any reflected acoustic intelligence is incident only on those hydrophones that face its path. The output of each of these hydrophones is connected through their individual preamplifiers to corresponding stator segments on both the audio and the video scanning switches. The scanning switches do not effect a direct contact but utilize a capacitive connection. The stator plates connect to their corresponding hydrophone units, and the rotor plates-18 in number-connect to a lag line to form the acoustic beam. Scanning switches are needed to interpolate bearings; otherwise the bearing could be obtained only in steps of 7.5° and the accuracy of the equipment would be impaired.

The video scanning switch is driven at a continuous rate of 1,750 rpm. Geared at a 1-to-1 ratio with this switch is a control transformer that positions the electron beam of the cathode-ray tube so that it remains in synchronism with the true bearing of the scanning switch.

The rotor of this control transformer is excited by a d-c voltage that is varied linearly with time to produce a slowly expanding spiral sweep. The picture of the cathode-ray tube therefore indicates plan position with the ship at the center. The audio switch is identical to the video switch but is not continuously rotated. The rotor of the audio switch is positioned by a servo system that is controlled from the console. This servo system drives another control transformer, which feeds the bearing information back to the cursor line. The cursor line appears on the screen of the cathode-ray tube during the transmission period and indicates the true bearing to which the audio channel is trained. Two scanning switches are needed because the video switch is rotated so rapidly that audio signals from it occupy too small a time duration to be heard. The inputs to the scanning switches are parallel, but their

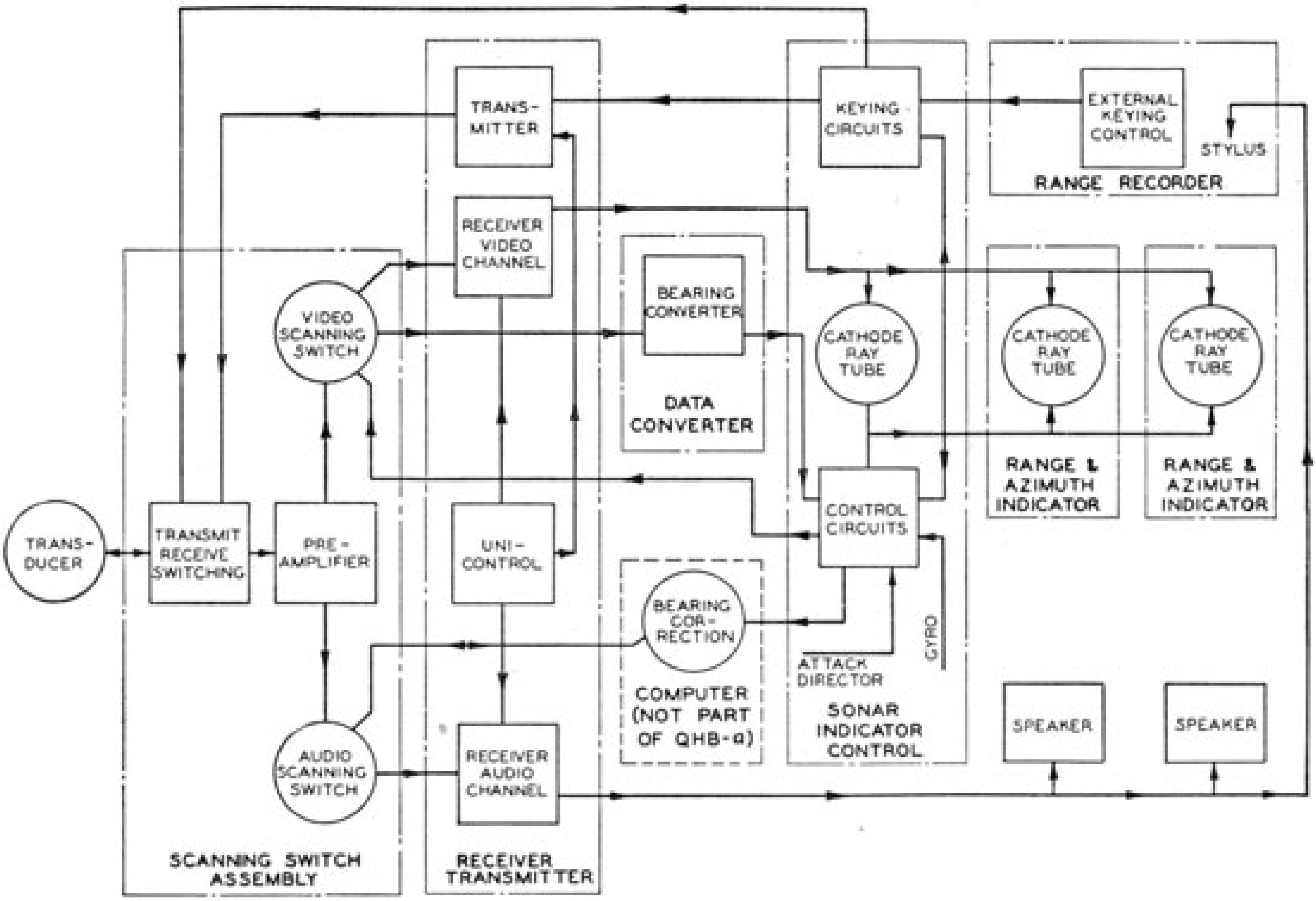


Figure 6-7. -Block diagram of the QHB-a system.

outputs are separate and each supplies a receiver. The circuits just discussed are called the *directional-sensitivity circuits*.

The transmitter is a conventional pulse-type amplifier.

A block diagram, figure 6-7, illustrates the over-all operation of the QHB-a equipment. Transmission is initiated by a keying pulse, which originates in the keying-pulse generator circuit of the sonar indicator control and which either functions automatically or is triggered by a range recorder. This pulse operates the relay for the transmit-receive switching and produces the transmitted indicates direction of the ship's stern. The bearing cursor appears automatically on the screen of the cathode-ray tubes at a bearing that corresponds to the direction in which the acoustic beam of the audio scanning switch is trained. Cursor time is confined normally to a portion of the transmission interval but may be extended when the OKA-1 sound range recorder is in operation. When slewing, the bearing cursor automatically appears continuously. The stern-line indicator appears only during the sweep interval and progresses with the sweep from a small arbitrary range to the maximum range use.

Description of Components

Transducer. -The transducer is the underwater element that performs the fundamental function of reciprocal conversion of acoustic energy into electric energy. It is a simple cylinder, as shown in figure 6-8.

power at a frequency derived from the unicontrol-oscillator system of the receiver. This transmitting electric power is transformed into acoustic power by the transducer and transmitted simultaneously in all directions. At the end of the transmission the keying relay restores the circuits to the condition for receiving, and the transducer then acts as a hydrophone and produces electric signals from acoustic reflections or noise sources.

The video scanning switch, rotating continuously at 1,750 rpm, produces a signal voltage whenever its acoustic beam sweeps past an echo signal. This voltage is then delivered to the video channel of the receiver, a conventional superheterodyne, the rectified output of which supplies brightening signals for the grids of cathode-ray tubes. The beam deflection in these tubes is synchronized with the video scanning switch so that the brightening occurs at the correct indicator bearing. Approximate range is shown by causing the axial deflection of the beam to increase at an appropriate rate with respect to time. This procedure produces a slowly expanding spiral sweep.

The audio scanning switch, which can be positioned by the training control, receives echo signals from a particular bearing and delivers them to the audio channel of the receiver. This channel also is a conventional superheterodyne with a beat-frequency oscillator for producing audio notes from the ultrasonic echo signals. The output of this channel supplies echo signals to the loudspeakers and the range recorder.

The construction and harnessing of the units are such that the mechanical Q is approximately 12. When the transducer is connected to the system load, the effective Q is reduced to approximately

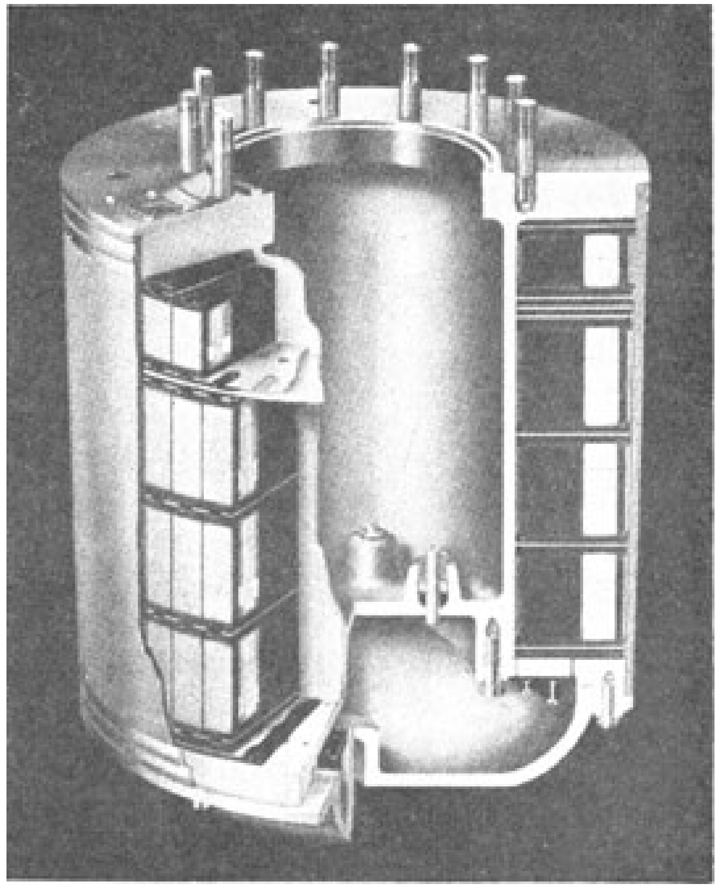


Figure 6-8. -Cut-away view of the QHB-a transducer.

In addition to the echo scanning, the video presentation includes (1) a radial solid line, or bearing cursor, and (2) a radial dotted line, which

8 ½, thus providing a 3-kc operating frequency band.

The transducer is composed of 48 transducer units mounted radially in the transducer. A cut-away view of a single unit is shown in figure 6-9.

Directly above the array of 48 transducer units is a similar ring of 48 smaller units, each of which is but 1 3/4 inches high. These units are series-connected and are employed only for transmission when echo ranging on deep, nearby targets. This manner of transmission is called *maintenance of close contact* (MCC). The short vertical dimension of the units provides a broad vertical transmission pattern, assuring that sound energy reaches a target at a large depth angle. The vertical response of the main portion of the transducer has a gain of approximately 11 decibels over a nondirectional radiator. The two-way loss in echo ranging with this pattern makes contact with a deep target unlikely, so the MCC units are keyed with the main units in order to distort the beam into a broad vertical pattern.

Scanning-switch assembly. -The scanning-switch assembly (figure 6-10) is concerned primarily with the preamplification of the 48 signals from the transducer and the formation of acoustic beams from these signals. It contains the send-receive switching provisions and the means for changing over from normal transmission to MCC transmission.

The preamplifier unit contains 48 identical resistance-coupled amplifiers, each of which is associated with a specific transducer unit. Plate and filament supplies are obtained from the power-supply chassis. Each amplifier consists of an input transformer and a twin triode, 6SL7, with associated capacitors and resistors. The output of each amplifier is at low impedance, and permits connection to the scanning switch without use of a twisted pair as required in the input.

The video scanning switch is identical to the audio scanning switch in all respects except the method of rotation of the rotor. This video switch is mounted directly below the audio scanning switch. Signal connections to the video switch are direct from the audio-switch terminal board. Instead of being driven by a servo, the rotor shaft is continuously driven at a 2-to-1 reduction by a capacitor-type induction motor that is rated at

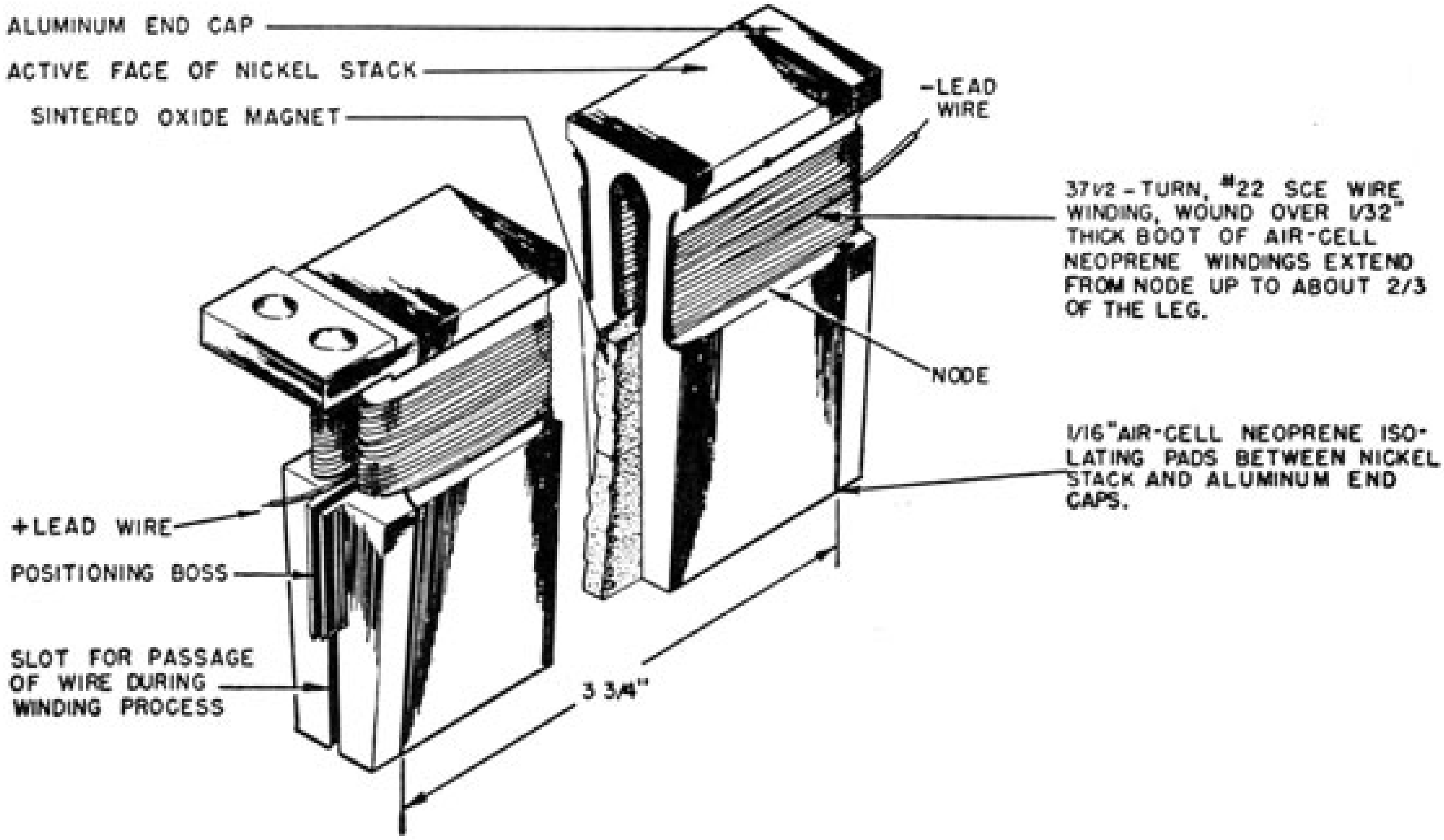


Figure 6-9. -Cut-away view of QHB-a transducer unit.

The range and bearing of an echo thus may be identified by the appearance of a bright arc on the screen of the cathode-ray tube.

It may be noted in passing that the angular speed of the scanning-switch rotor and the pulse length of the transmitter are such that the rotor is "trained" on a possible returning echo at least once during the time the echo is incident on the transducer. Shorter transmitter pulses would possibly result in failure to detect echoes because all sound energy might return during the time the rotor was trained in other directions. The pulse length of the equipment is 35 milliseconds, or 1/28.5 seconds, the returning echo has the same time duration, and as the angular speed of the scanning switch is slightly less than 30 revolutions per second, any

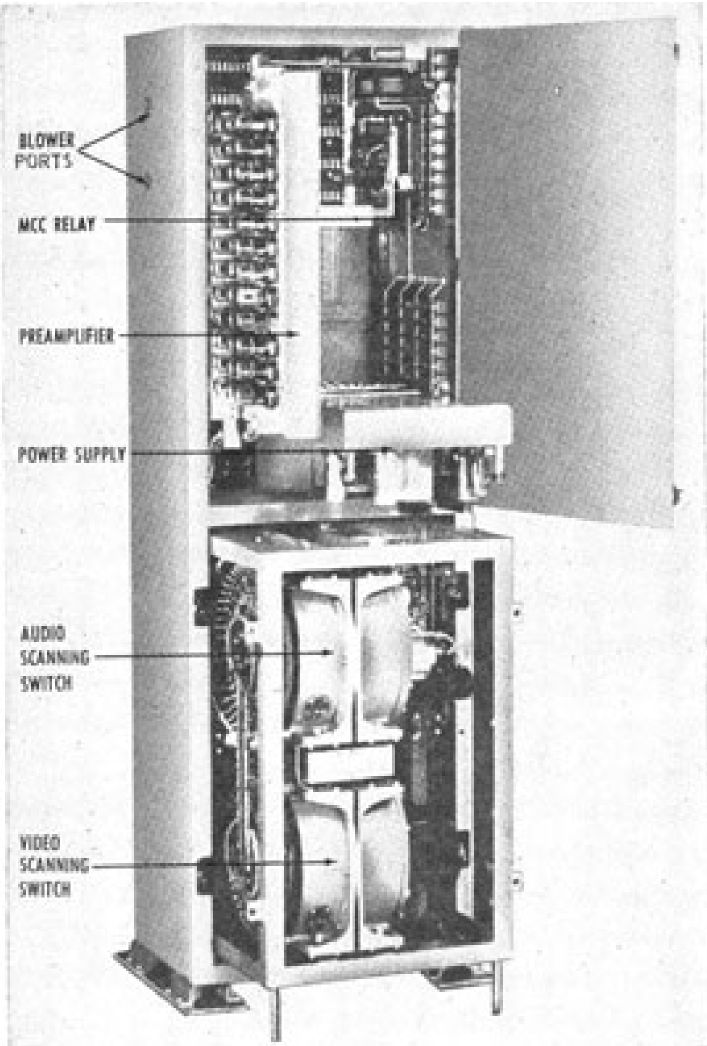


Figure 6-10. -QHB-a scanning-switch assembly, showing interior.

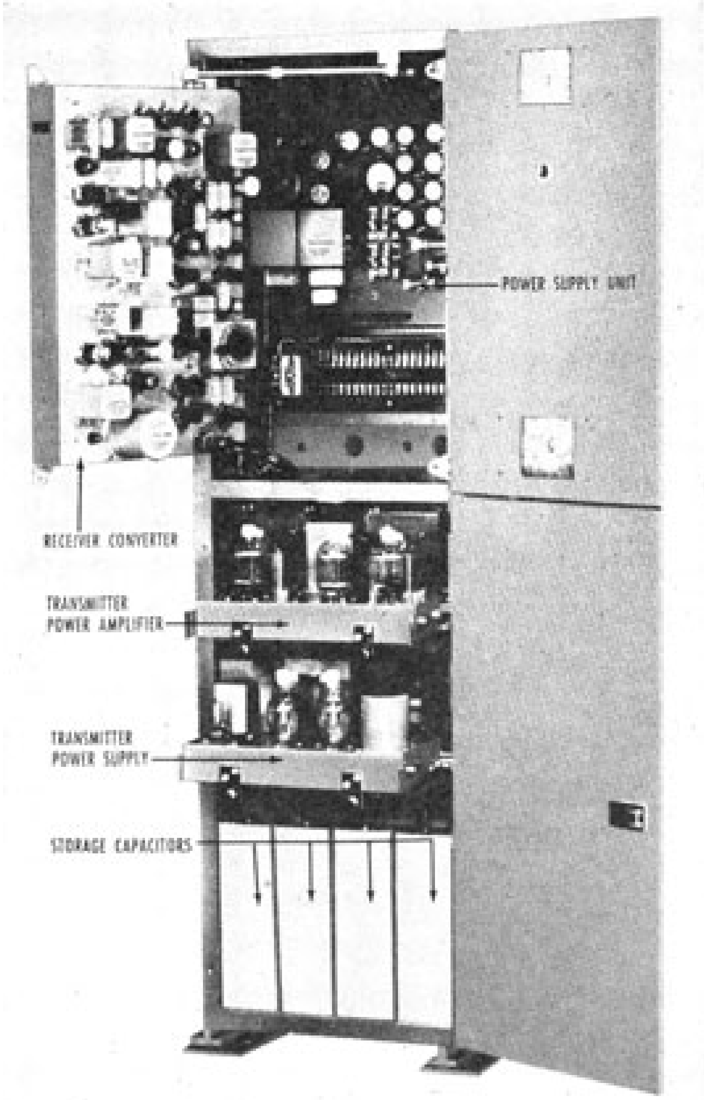


Figure 6-11. -QHB-a sonar receiver-transmitter cabinet, showing interior.

3,500 rpm and 1/20 hp. The drive is accomplished through helical gears to ensure smoothness of rotation and reduction of noise. A 5HCT control transformer is driven at a 1-to-1 ratio by the scanning-switch shaft. The rotor of this control transformer is excited by a direct current that is proportional to the sound range. Therefore, a 3-phase a-c voltage is induced in the stator at slightly less than 30 cps.

The magnitude of the 3-phase voltage is proportional to the range, and its phase relation is constant with the instantaneous angular position of the scanning-switch rotor. This polyphase signal is connected to the deflection coils of a cathode-ray tube. The plan position picture is fairly accurate if the beam of the tube is precisely

synchronized in angle with the scanning switch rotor and in radius with range. The output of the scanning-switch rotor is connected through suitable amplifiers to the grid of the indicator cathode-ray tube.

returning sonar intelligence is incident on the transducer during a scanning cycle.

Receiver-transmitter unit.-The receiver-transmitter unit (figure 6-11) contains a dual-channel receiver, which amplifies the signal from the two scanning switches to a level that is suitable for operation of the video and audio indicators of the system. It also contains the complete impulse-type transmitter, which provides the high-level electric signal employed in echo ranging. On the outside, this unit is a simple box without any controls.

When the upper door of the cabinet is open, only the receiver-converter is accessible. When the lower door is open, the complete transmitter is accessible. In the bottom of the cabinet are mounted four large oil-paper capacitors, which constitute the energy storage for the high-power pulse. On the cabinet structure directly above the capacitors are mounted two interlock switches operated by the door. One disconnects the 3,700-volt power supply, and the other short-circuits the capacitor bank, thereby minimizing the hazard from extremely high potentials.

Directly above the storage capacitors is a drawer that contains the high-voltage supply for the transmitter. Directly above the power-supply chassis is another drawer that contains the transmitter-power amplifier. To prevent possible hazard to maintenance personnel, the upper portion of the cabinet is separated mechanically from the transmitter section by an expanded metal

thereby governing the power-output level of the transmitter; and (4) *target-doppler nullification on-off switch*, which completely disables the target-doppler nullification circuit without disabling own-doppler nullification. All the adjustments described here are of the screw-driver locking type.

The transmitter power amplifier is contained in a drawer in the upper portion of the transmitter section of the cabinet. In it are mounted 3 type-715C beam-power tetrodes, the filament transformer, the input transformer from the converter, and the output transformer tuning capacitors.

grill. The receiver-converter unit is the vertical chassis mounted in the front part of the upper portion of the cabinet. The chassis contains the complete twin-channel amplifier plus the converter section, which provides the output-frequency signal at a suitable level for driving the power amplifier. The tuning range provided is from 22 to 29 kc. The transducer that is furnished with the equipment is useful only over the range of from 24 to 27 kc.

Other operational controls on the chassis are (1) *audio gain control*, which provides for adjustment of the level of the audio output with respect to the video; (2) *doppler-nullifier gain control*, which provides for the "stiffness" of response of the doppler-nullifier circuit; (3) *converter gain control*, which adjusts the level of the signal to the transmitter,

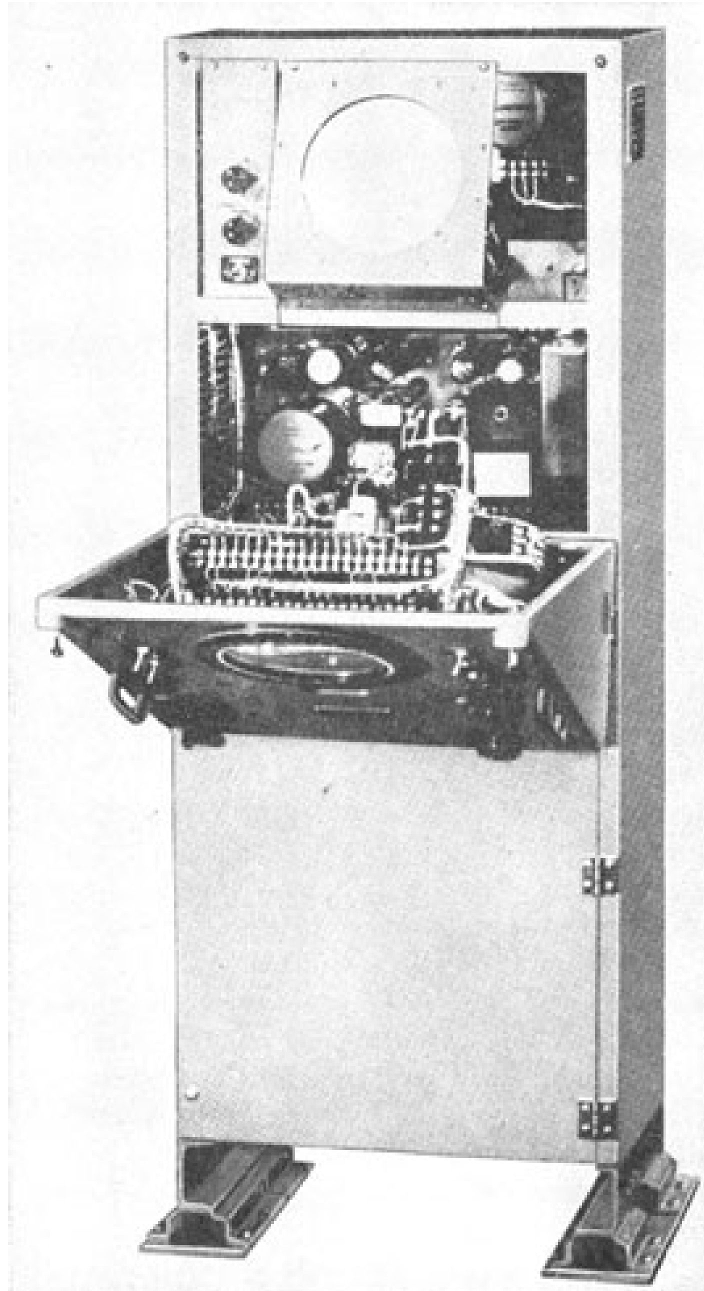


Figure 6-12. -QHB-a sonar indicator control, showing interior.

The unit is extremely simple and contains a minimum of internal components. No adjustments are involved in the circuit.

Electrically, the function of the unit is to accept from the converter an r-f signal pulse of relatively low power. The unit amplifies this pulse to a power level of approximately 7 kilowatts maximum, which attenuates approximately 3 ½ db during the pulse time. This attenuation is characteristic of a pulse-type plate supply.

The transmitter power supply has the single function of charging the storage capacitors during the interval following the echo-ranging transmitting pulse. This power supply consists of a large power transformer, the primary of which is in series with a current-limiting reactor; the secondary supplies the plates of two type-866A rectifier tubes. The cathodes of these tubes are connected directly to the storage capacitors and to a resistance network, which serves as a bleeder. These resistors are mounted on the underside of the chassis, and a portion of the combination is paralleled by the voltmeter on the power-supply plate.

Sonar indicator control. -The sonar indicator-control unit (figure 6-12) is the sonarman's station and contains all the operating controls.

In the upper portion of the cabinet is mounted the assembly of the cathode-ray tube. Immediately to the left of this assembly is a small control panel. On it a toggle switch provides for selection of peak or band filter in echo-ranging operations. In the *peak* position it provides target-doppler nullification if this feature has not been disabled by the switch in the receiver. It is undesirable to echo range with this switch in the *peak* position if the target-doppler nullifier circuit has been disabled in the receiver, because the peak filter is so sharp that a target with appreciable doppler

Remote indicator. -The remote range and azimuth indicator is essentially a cathode-ray tube repeater. Associated with it is a loudspeaker. This remote installation gives a complete duplication of the visual and audible indications available to the sonarman. The design of the QHB-a is such that two range and azimuth indicators may be installed with each equipment.

The upper portion of the front panel of the remote indicator contains a circular opening covered by amber filter glass through which the cathode-ray tube is viewed. Surrounding this opening is an azimuth ring identical with that on the sonar-indicator control. Below the panel to the right are various controls-the threshold-intensity and focus adjustments of the cathode-ray tube, the video-signal level adjustment, and the loudspeaker volume control. These controls make possible the complete and independent adjustment of visual and audible indications at the remote station as long as the sonarman operates the equipment at a reasonable level.

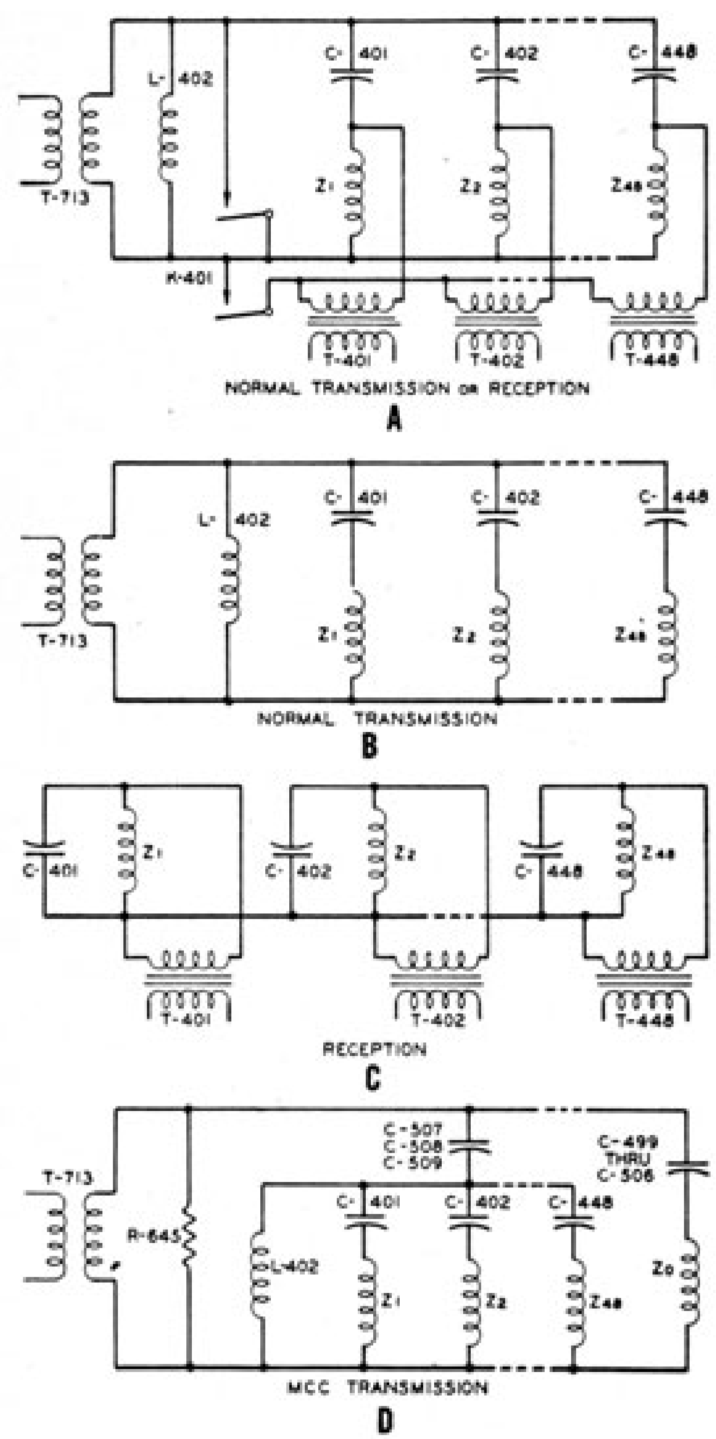
Transmit-Receive Switching

The 48-transducer unit, the transmitting capacitors, and the signal transformers from the connection for transmission to those for reception are switched by two contacts on keying relay K401. For transmission, each capacitor should be in series with a transducer unit, and the preamplifier input transformers should be connected so that the voltage developed across each of them by the transmission pulse is small. For reception, each transformer should be connected to its respective transducer unit, which is paralleled by a capacitor.

These switching connections are shown in condensed form in figure 6-13, A, where the connections for transmission occur when the relay contacts are open and where the connections for reception occur when these contacts are closed. The

provides very little audio indication. In the *band* position an RC band filter is inserted, and target-doppler nullification is eliminated. On this same control panel are two potentiometers, one governing the threshold signal of the cathode-ray tube and the other governing the intensity of the electronic bearing line or cursor. In the rear of the upper portion of the cabinet is mounted the second-anode supply of the cathode-ray tube.

48 transmitting capacitors, C401 through C448, are connected in series with their respective transducer unit, Z1 through Z48. One terminal of each of the 48 input transformers also is connected to the corresponding transducer unit; the other terminals are connected together during transmission and are separated from ground by keying relay K401.



and transformers. As a result, the circuit for normal transmission can be shown functionally, as in figure 6-13, B.

Each transducer unit, Z1 through Z48, including 40 feet of cable, has a nominal impedance of 58+ j82 ohms at center frequency, and the series capacitors, C401 through C448, produce a total effective impedance for each unit of 58-j400 ohms. The 48 parallel circuits then present, at 25.5 kc, an impedance of 1.21-j8.35 ohms. The tuning inductor, L402, in parallel with this combination, has a reactance of 8.8 ohms. These values produce a load that is equivalent to approximately 50 ohms on the output transformer, T713.

Figure 6-13, C, shows the equivalent circuits for reception. The impedance of the signal source, which consists of a transducer unit paralleled by one capacitor, becomes equivalent to 81+j86 ohms. Hence, the transformers are designed to reflect the input impedance of each preamplifier circuit and produce a primary impedance of 81-j86 ohms, which results in a conjugate impedance match to the transducer circuit and the greatest transfer of energy.

To provide a broader vertical beam pattern and to reduce the transmitted acoustic intensity in the horizontal plane, a condition called *MCC transmission* can be established for purposes of maintaining close contact.

D

Figure 6-13 -QHB-a transmit-receive switching.

If the capacitors, transducer units, and transformer input impedances were identical in each of the 48 circuits, the a-c potentials applied to the transformers would be equal and the return circuit from the transformer common connection would be open. This condition would result in no voltage across the transformer primaries. Hence, the voltages existing during transmission in any of the 48 transformers have a value that depends on the inequalities of the 48 capacitors, transducer unit,

In addition to the main units, the series-connected ring of 48 short units-located in the upper end of the transducer and called the *MCC ring*-is used to transmit with a broad vertical pattern. This ring is not involved in reception. Some power is supplied to the remaining transducer unit, with phase and amplitude relations between these units and the MCC ring sufficient to cancel the transmitted intensity along the horizontal axis. This cancellation reduces surface-reverberation effects. The circuit with the MCC connections is shown in figure 6-13, D.

Directional Sensitivity Circuits

General -The directional sensitivity circuits cover the signal-receiving function from the transducer, through the preamplifiers and scanning switches to the receiver.

In the receiving condition the de-energized keying relay connects the 48 transmitting capacitors in parallel with their respective transducer units.

The keying relay also bridges the primaries of the 48 preamplifier input transformers across the transducer units. Each of these transformers provides a conjugate impedance match between (1) the transducer-unit circuit, including the capacitor, and (2) the input impedance of the preamplifier tube. Because the 48 preamplifier circuits are identical, this discussion deals only with one of them. The signal from the transformer is connected to the control grid of the left side of the twin triode, which is connected as a voltage amplifier with a gain of approximately 20. The output is connected to the right side of the twin triode, which is operated as a cathode-follower. Thus, it is a low-impedance source for transmission to its corresponding segment on the stator plate of the audio and video scanning

The usual beam pattern for this type of transducer consists of a major lobe, accompanied on each side by minor lobes that decrease in sensitivity as the angle from the main lobe increases. Altering the relative amplitude of the side-lobe signals (shading) can result in a reduction of the level of the more adjacent minor lobes at the expense of increasing the very small lobes, which are at a large angle from the main beam. The optimum signal-to-noise ratio occurs for such shading when the minor lobes have all been brought to the same level, and consequently the main beam is widened only slightly. Design engineers select the proper fraction of each of the signals that are to be phased and added so as to provide this shading.

Lag-line phasing and shading. -The beam-pattern

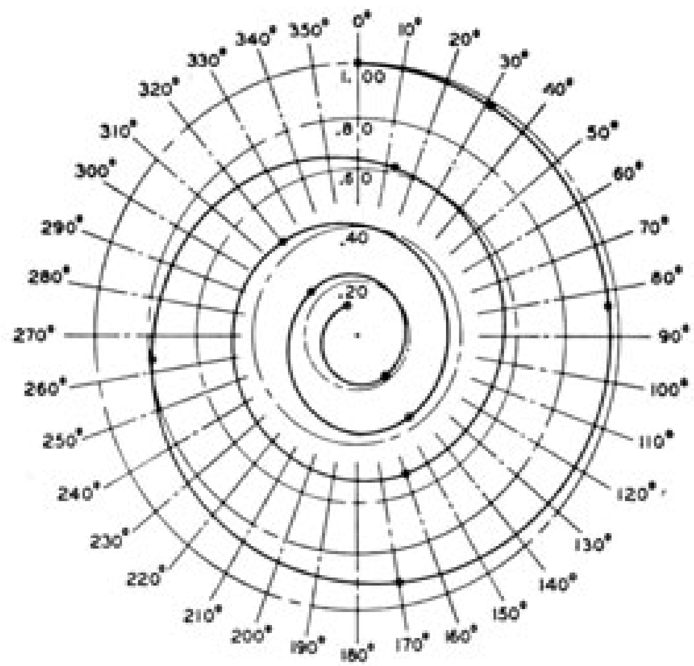
switches. Similarly, the remaining 47 preamplifiers deliver the amplified transducer-unit signals to the other 47 capacitor segments on each scanning-switch stator.

Beam-pattern formation and rotation. -One function of the equipment is to produce electrically an acoustic pattern, continuously rotatable through 360°, using a fixed cylindrical array of hydrophones or transducer units. This function is accomplished by two devices-(1) the electric circuits necessary to produce the beam pattern and (2) the electro-mechanical means for rotating this pattern.

The production of an optimum beam pattern from a fixed array of radiators or receptors is a fairly well known art and consists either in (1) choosing amplitudes and phases for the currents in the radiators or (2) modifying the voltages from the individual receptors-depending on the geometry of the array. The phasing requirements are imposed because this array of transducer units is cylindrical. Therefore, the signals received by each unit from a plane sound wave, unlike the signals that exist in a plane-faced transducer, differ in phase in proportion to their physical displacement. The total voltage from a group of units facing the sound source is a maximum when all the signals have been shifted so as to be in phase with one another. The resulting beam pattern is similar to that of a plane-faced transducer of approximately equivalent dimensions. This phasing requirement is accomplished by the use of a linear phase-shift "lag line" in order that the phasing may remain correct when the frequency is changed.

formation can be analyzed by (1) inspection of the voltages produced by the transducer units, (2) the required phasing, and (3) the choice of shading.

The first of these methods, involving a single-unit pattern, may be presented in several ways, one of which is shown in figure 6-14. This illustration shows that the total lagging phase shift for the signal from unit 1 is at least 680° in order to bring it into phase with the signal from unit 8. The



ELEMENT	1	2	3	4	5	6	7	8	9	10	11	12
AMPLITUDE	1.00	0.99	0.97	0.92	0.79	0.62	0.54	0.45	0.36	0.25	0.18	0.12
PHASE	0°	31°	84°	170°	264°	373°	520°	680°	867°	1031°	1224°	1422°

Figure 6-14 -QHB-a transducer-unit voltages.

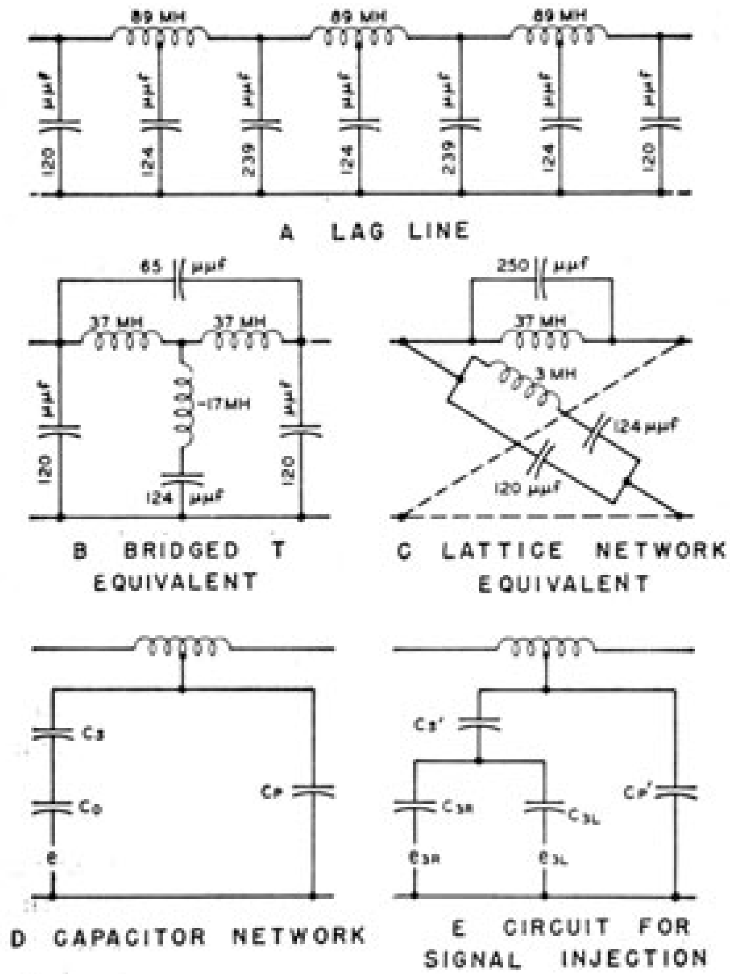


Figure 6-15 -QHB-a lag line and equivalent circuits.

phase shift for the voltage from unit 2 is 31° less than that from unit 1.

To accomplish the necessary phase shifts, the simplest lag line is a uniform line of as many sections as required to match the desired angles with whole or half sections. One limitation is that the phase shift per section must be kept below approximately 60° in order to approach linear phase shift with frequency. The final lag line as designed has a phase shift of 52° per full section at 26 kc and consists of 14 sections. The physical arrangements for three sections are shown in figure 6-15, A. The proper choice of entry points for the signal voltages allows a good approximation to the required phase shifts. The electrical equivalent of this circuit for a single section is shown as a bridged-T network in figure

A method of injecting the signal voltages from low-impedance generators has been devised. This method does not impose any loading of the lag line or mismatching at any point, which would result in standing waves of voltage on the line. Because the scanning switch must introduce these voltages through the capacitance of the segments on the stator and rotor plates, this capacitance is made a part of a network of three capacitances, through which is injected a voltage (figure 6-15, D).

Two units of the scanning-switch circuit that are symmetrically disposed about the center of the beam-forming network (such as 3R and 3L in figure 6-17), introduce signals at the same point on the lag line. The complete circuit for signal injection is shown in figure 6-15, E. The values are related because the total capacitance must equal the value required (1) by the lag line at the point of signal injection and (2) by the desired fraction of the signal voltage that is to appear on the lag line.

The optimum beam pattern requires an attenuation or shading of the transducer-unit voltages. The desired total attenuations are shown in figure 6-16, with the attenuation already present, that is due to the single-unit pattern. The ratio of the single-unit pattern to the total attenuation for any unit determines the attenuation that must be introduced by the beam-forming network. This

6-15, B, and as the equivalent lattice network used in design in figure 6-15, C. This circuit results in a characteristic impedance of 16,300 ohms at 26 kc.

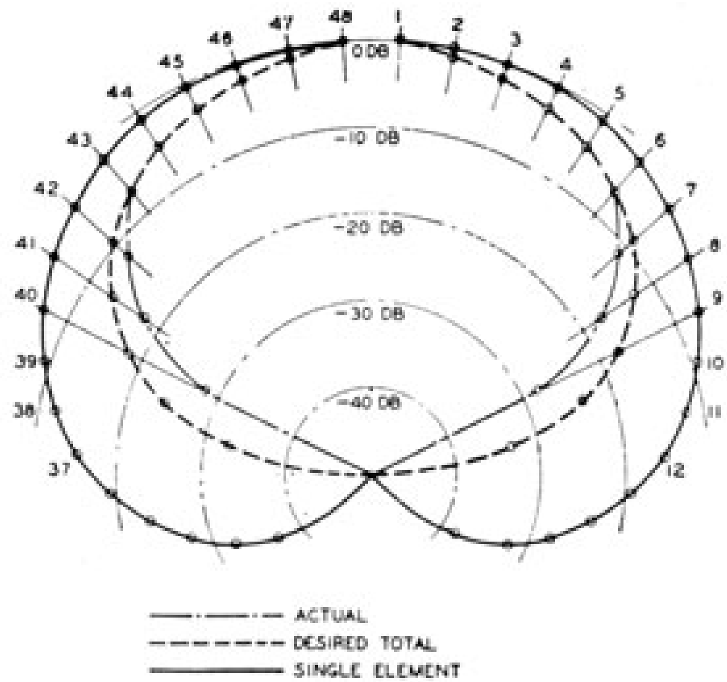


Figure 6-16 -Shading curves.

attenuation is accomplished by the choice of capacitance values shown in figure 6-15, E. The combination of these circuits results in a complete scanning-switch circuit, illustrated in figure 6-17. Because the phase shift from unit 7 to unit 8, and from unit 8 to unit 9, is approximately 180°, a voltage also is used from unit 9 and is introduced into the same point as that from unit 7.

A typical resultant beam pattern has (1) a major lobe 11° wide at a level that is 6 db below the peak sensitivity and (2) minor lobes that are at least 25 db below the same reference level.

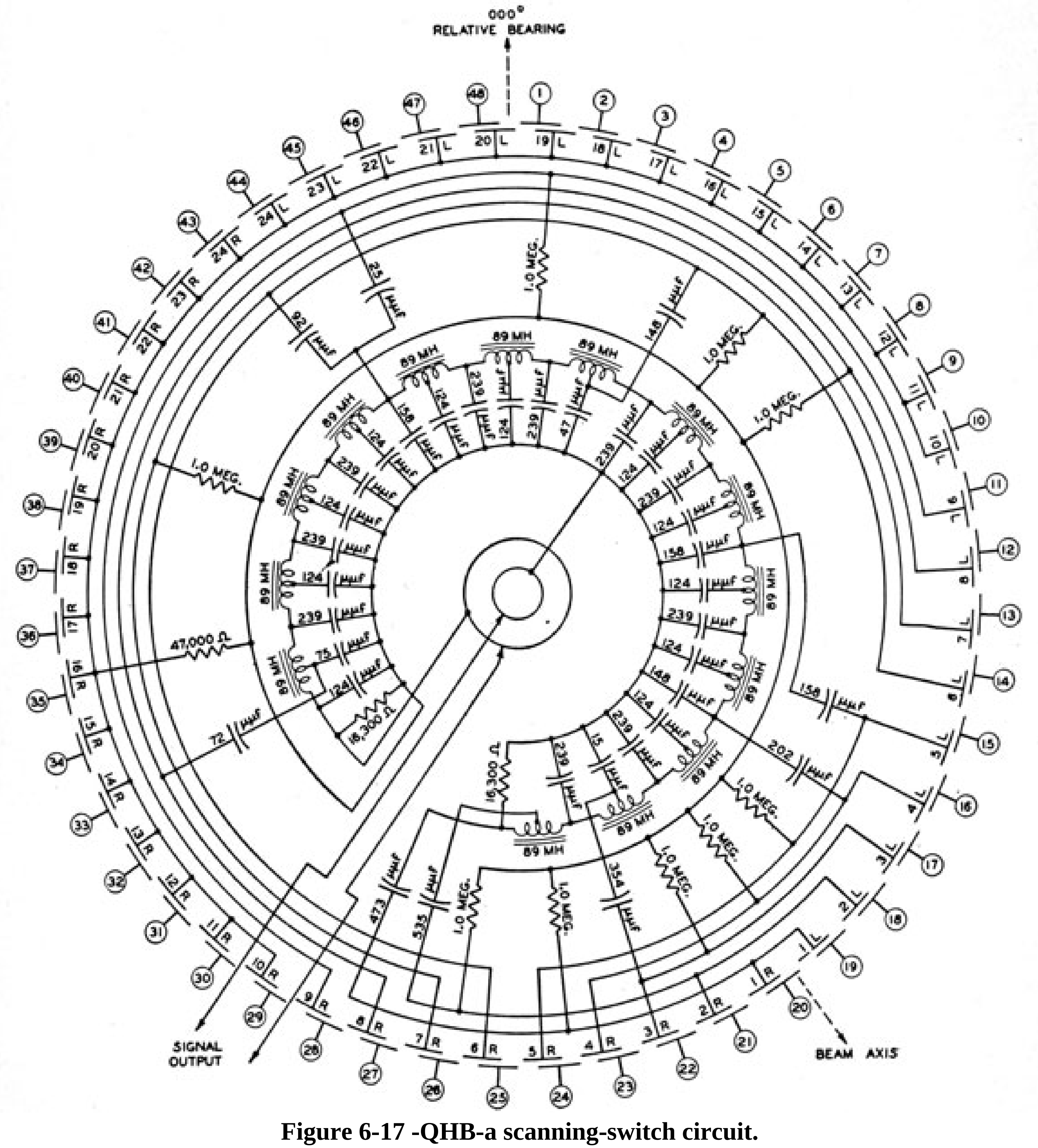


Figure 6-17 -QHB-a scanning-switch circuit.

Scanning switches. -The video and audio switches are identical both structurally and electrically. Structurally, they each consist of two large cast-iron cups accurately doweled and bolted together at their open ends. At the left end of the assembly is a circular terminal board, to which are connected the 48 output leads from the preamplifiers. These leads are connected directly by short conducting loops to pins that protrude through the left-hand surface of the scanning switch proper. The pins are insulated by rubber grommets.

A glass disk, 11 inches in diameter and 7/8 of an inch thick, is bolted securely to the machined surface of the left-end bell. The outer plane surface of the disk is coated with silver and scribed with 48 radial divisions, which form separate conducting segments. After 48 holes have been drilled completely through the segments and the glass, they are metalized. In this way the segments are connected electrically to the back of the hole, where connection pins are soldered, thereby providing a complete circuit from any specific preamplifier to a corresponding segment. Each of the rotors consists of a glass plate (identical to the stator plate), which is secured to a large steel hub that has been shrunk on the rotor shaft. With the rotor shaft mounted in place, there is an air gap of approximately 0.004 inch between the glass plates.

With the segments of both rotor and stator lined up, 48 equal capacitors of from 80 to 100 micromicrofarads are formed. These capacitors constitute the means for connecting the outputs of the preamplifiers to the electrical network carried on the rotor assembly. This network provides the proper phasing and attenuation of the signals from any 18 consecutive stator elements in order to form an acoustic beam. The network is mounted in a cast-aluminum can, and the output connections are carried to suitable slip rings these switches are parallel-connected, and their outputs are connected to individual receivers-one for video and the other for audio presentation. The system lined-up for the receiving functions is as follows:

1. Each transducer unit connects through its own preamplifier to a stator segment on the audio scanning switch and to the corresponding stator segment on the video scanning switch.

2. The output of the video scanning switch connects to the input of the video receiver, the rectified output of which is used as the brightening signal for the control grids of the cathode-ray tubes.

3. The output of the audio scanning switch connects to the input of the audio receiver, the output of which is used to drive the loudspeakers and to mark the tactical range recorder.

The azimuth angle of the video scanning-switch rotor at any instant is indicated on the cathode-ray tube by a 5HCT control transformer. This control transformer is used as a 3-phase sweep generator-a unique employment of a synchro.

In the usual synchro system the rotor is energized by a single-phase a-c voltage. Therefore, the stator coils remain in time phase but vary in magnitude, depending on the angle between the axis of the magnetic field and the axis of each stator coil. When these voltages are connected to a synchro receiver they duplicate the magnetic axis of the transmitter. The 5HCT synchro used to generate the sweep voltage is excited by a direct current, the magnitude of which is proportional to the range of the active volume at any instant. The output therefore is a true 3-phase voltage with the peak magnitude of the voltage in each phase increasing with time. If the excitation remained at a constant value this arrangement would be the same as that in any 3-phase generator the rotor of which is d-c-excited and

mounted on the rotor shaft. Carbon-silver brush members inserted through the rear wall of the right-end bell engage these slip rings, thus making accessible the electrical-signal equivalent to the acoustic information that is obtained at any instant.

Because of the necessary high rate of rotation of the video switch the signal from an echo is not sufficiently long for audio presentation. Therefore, two switches are needed. The inputs of rotated in a 3-phase stator.

The rotor of the 5HCT synchro that is used to generate the sweep voltage is geared at a 1-to-1 ratio to the video scanning-switch rotor. When the system is in motion a 3-phase voltage proportional to range and synchronized in bearing is available at the stator terminals. Because the rotational speed is about 1,750 rpm, the output

frequency is slightly less than 30 cps. This polyphase signal provides relative bearing of the video beam in the deck plane.

For conversion to true bearing with stabilization, the signal is taken to the data converter and is applied to the stator terminals of a 5SCT synchro, which is used as a polyphase phase-shifter. The rotor of the 5SCT is positioned primarily by the ship's gyro order with a stabilizing component related to azimuth sonar train. The output at the rotor terminals of the 5SCT synchro is therefore a 3-phase true bearing sweep signal that is stabilized in a horizontal plane, with respect to the line of sight. The three components must be amplified by three identical feedback amplifiers so as to provide sufficient signal for the deflection coils of the cathode-ray tube. These coils are the stator coils of a 5SCT synchro.

Depth-Determining Equipment

MODEL QDA

General

The model QDA depth-determining equipment is an ultrasonic echo-ranging equipment operating in the frequency band of 50 to 60 kc. It is primarily an attack instrument and is installed in conjunction with an OKA-1 sonar resolver and an azimuth sonar equipment, which may be either the QHB-a or the QGB. The differences between the QDA equipment and a standard azimuth search equipment lie chiefly in the transducer and the recording mechanism.

As with ordinary echo ranging, range is *Vz*. The stylus moves at a rate corresponding to the slope of the sound beam. If the beam is steeply inclined the stylus moves rapidly from left to right; if the beam is nearly horizontal the stylus moves slowly. The ping is transmitted just as the stylus moves away from its zero position, and the stylus marks the recorder paper at the instant the echo returns.

For a distant target the depression angle is small, and correspondingly, the stylus moves slowly but for a relatively long period before the echo mark is recorded. For a nearby target the depression angle is relatively large, and as a result the stylus moves rapidly for a short period before the echo returns. In both cases the echo is recorded at the same distance

determined by the ping-to-echo time lapse and the velocity of sound. For determining the depression angle, the QDA transducer, which has a sharp beam in the vertical plane, is pivoted on an athwartship axis to permit the beam to be tilted to any necessary depression angle. In depth search, pings are sent with the beam tilted at various angles. When the beam is directed toward a target, echoes are detected if the target is within range, and the tilt of the beam at the time such echoes are received corresponds approximately to the target depression angle *Eq*. After depth contact with a target, the alignment of the beam with the target depression can be indicated more accurately by a *depression deviation indicator* (DDI), which is analogous to the bearing deviation indicator employed in azimuth sonar systems.

Depth is indicated automatically by a depth recorder, which is similar in principle and construction to the tactical range recorder and the indicator range recorder. In the depth recorder, however, the stylus travels at a speed proportional to the sine of the target depression angle, which is the vertical component of the velocity of sound, from the starting position of the stylus. A linear depth scale, reading in feet, extends across the recorder chart. The stylus speed is controlled by the OKA-1 resolver, but the basis of this speed is determined by the QDA beam depression angle and the velocity of sound in water. When the bypass switch is in the *search* position the azimuth transducer only is stabilized. When the bypass switch is in the *attack* position both the azimuth and the QDA transducers are stabilized.

Keying and Controlling the Recorders

The keying interval for both the azimuth and the QDA echo-ranging systems is controlled by the sound-range recorder, a unit of the OKA-1 equipment. When the fly-back contacts of the sound-range recorder are closed, the action of the keying circuit in the azimuth equipment and of a similar circuit in the OKA-1 resolver are both initiated. The latter circuit controls the depth recorder and keys the QDA. These timing circuits cause the stylus clutch to release in the

indicator range recorder of the azimuth equipment, and of the depth recorder so as to allow the respective styli to fly back and dwell for a brief period. The clutches are then re-energized, and almost simultaneously the two equipments are keyed.

The general arrangement and sequence of operations of the units of the different equipments are shown in the block diagram of figure 6-18.

If the depth recorder reaches the end of its travel before the fly-back contacts of the sound-range equipped with a cursor, the position of which represents target depth below the transducer, *H'q*, which also is transmitted by synchro order to the predictor resolver circuit. From these two inputs the OKA-1 resolver computes the following:

1. Horizontal sound range, *Rhq*, which is transmitted to the horizontal recorder and to the remote indicators of the QDA equipment.

2. Sonar target depth, *Hq*, which also is transmitted to the remote indicators.

recorder are operated, the stylus flies back and dwells until the sound-range recorder initiates another keying cycle. When the sound-range recorder is turned off, the depth recorder is no longer slaved, and it controls its own keying cycle through the OKA-1 resolver timing circuit.

Horizontal Range and Computed Target Depression

The sound-range recorder is provided with an adjustable cursor, the position of which represents sound range, *Rq*. *Rq* is transmitted by synchro to the predictor resolver circuit of the OKA-1 resolver unit. The depth recorder likewise is

3. The computed sonar target depression, *cEtq*, which is transmitted to the tilt-control differential generators in the QDA console as an aid in maintaining contact.

The operator of the QDA console modifies the order *cEtq* by an adjustment of the tilt wheel under guidance of the DDI. The order introduced by the operator is known as adjustment of computed sonar target depression, *jEtq*. If the azimuth beam is centered on the target, as indicated on the DDI, the order leaving the console represents apparent depression of the acoustic path to the target, *Eq*. The order *Eq* is the sum

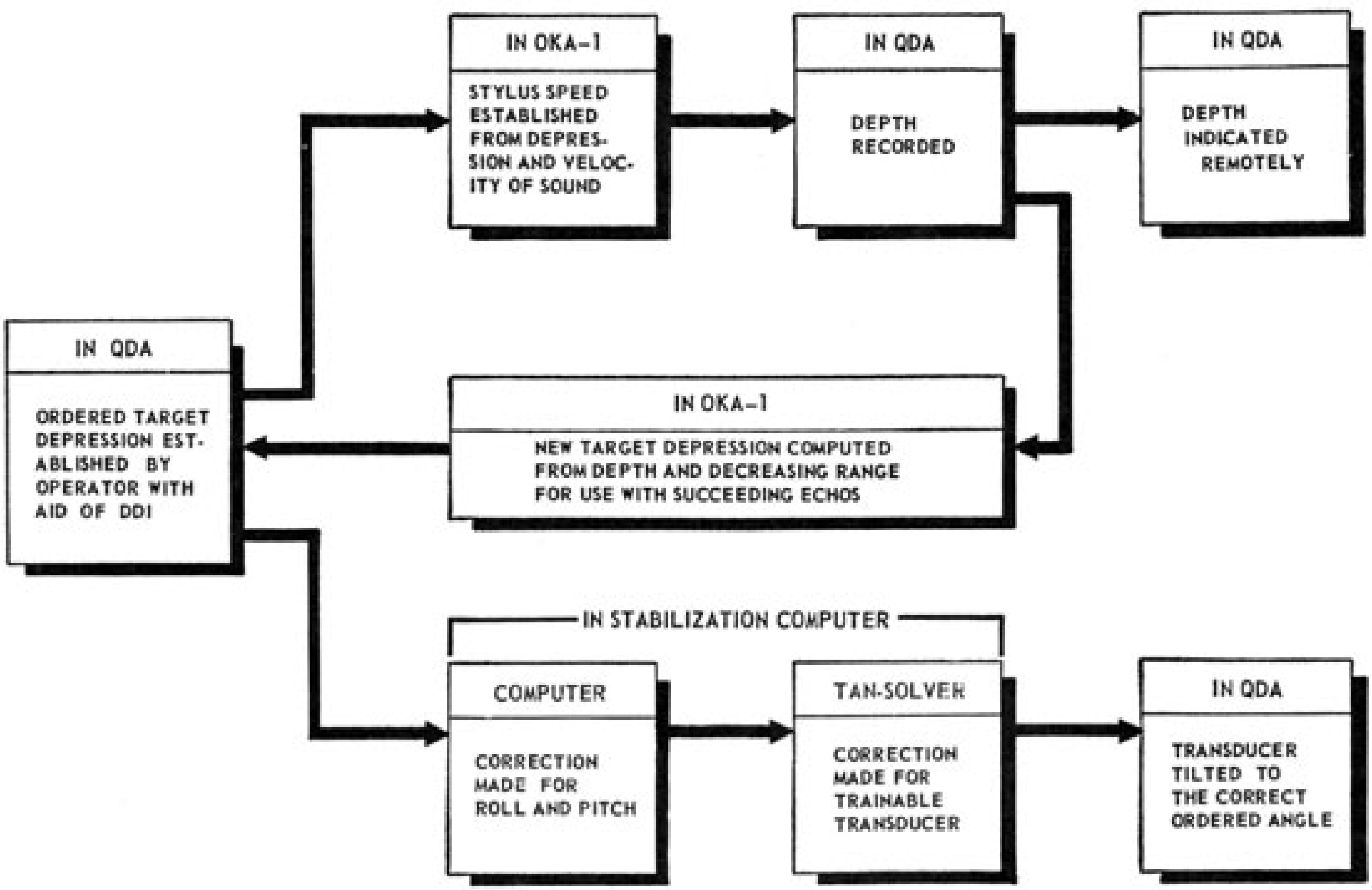


Figure 6-18. -Sequence of operations for determining target depth.

of *cEtq* and *jEtq*. This order is indicated on the console by a synchro that positions an indicator.

Correction for Bending of the Sound Beam

The order *Eq* is transmitted to the stabilization computer by way of the bypass switch and the Snell's law resolver circuit of the OKA-1 resolver unit. The purpose of the Snell's law resolver circuit is to establish the speed of the depth-recorder stylus so that its excursion rate is proportional to *Vz*. If the beam is vertical, the vertical component of the velocity of sound along the beam is simply the velocity of sound in water at the prevailing temperature. If the beam is horizontal, *Vz* is zero.

The Snell's law resolver computes and establishes *Vz* from the input information consisting of (1) the *Eq* order; (2) the velocity of sound, *Vo*, which is injected manually into the circuit; (3) the difference of velocity in the mixed layer and the refracting layer, *V*, also manually injected; and (4) the layer-depth timing introduced by the closure of the layer-depth contacts in the depth recorder at the appropriate point in the stylus excursion. The layer-depth contacts in the depth recorder are adjusted manually, and information for this setting of *Vo* and *V* in the Snell's law resolver and for *Vo* in the sound-range recorder are derived from bathythermograph information.

Tilt-Order Synchro Circuit

As has been stated, the depression angle order originates in the OKA-1 resolver, where it is computed and transmitted by synchro transmitters at 2 and 36 speed. For on-target conditions, this order is called *cEtq*. It is delivered to a pair of DG synchros to permit the operator to modify the computed order so as to center the beam on the target as indicated by the DDI. The signal introduced by the operator is *jEtq*. The output of *azimuth beam*, provided the computer is not bypassed. For on-target conditions, this angle is equal to *Eq*, the true depression at the transducer of the acoustic path to the target. In the absence of refraction or bending of the QDA beam, *Eq* is equal to *Etq*, the true depression of the target.

The true tilt angle of the transducer below the horizon is equal to the indicated angle, *Eq*, provided the azimuth transducer is trained dead ahead; but the true tilt angle exceeds the indicated angle by increasing amounts as the azimuth transducer is trained farther off the bow. When the bypass switch is in the *bypass* position, the tilt indicator shows the actual tilt of the QDA transducer with respect to the deck plane, and the transducer tilt is unaffected by the position of the azimuth beam.

The 2-speed output of the differential transmitters in the console is connected to the OKA-1 resolver to operate the Snell's law resolver, which controls the stylus speed of the depth recorder. This 2-speed output also is connected to the stabilization computer, which corrects it for the roll and pitch of the ship. This corrected order for on-target conditions is *E'q*, which represents the depression of the beam relative to the deck, measured in a plane through the line of sight perpendicular to the deck. *E'q* also is the order that would control the tilt of the transducer if the transducer were trained to the bearing of the target. For the QDA system, however, the *E'q* order is transmitted to the tangent solver of the computer, where it is converted into a transducer tilt order that causes the fan-shaped beam to pass through a target, which is at the bearing of the azimuth transducer. The transducer tilt order is *E'q's*.

E'q's is transmitted at 2 and 36 speed, by way of the tangent solver, to the control transformers on the tilt-control mechanism. The signals from these control transformers are connected to the tilt-control amplifier, which supplies the power to the tilt motor that drives the transducer to wipe out the signals and

the DG synchros is the sum of the two inputs and is equal to *Eq*, provided the azimuth and depth beams are on the target. A synchro receiver, which operates the tilt dial in the console, is connected to the output of the 2-speed DG to indicate this adjusted order.

It is important to understand the significance of the tilt dial indication. Whether or not the two acoustic beams are centered on the target, the indication represents the depression of the QDA *beam* below the *horizon along the bearing of the* thus bring the transducer to the ordered angle, *E'q's*.

The tilt of the QDA transducer is controlled by the following factors:

1. Factors controlling the depression order from the console. These factors are combined and then are indicated by the tilt indicator on the console.

a. The computed target-depression order is based on the position of the depth-recorder cursor and that of the cursor of the sound-range recorder. If either of these positions is changed, the order from the resolver to the QDA console changes.

b. The position of the two differential-transmitter rotors located in the console is controlled by the tilt wheel.

2. Factors controlling the correction of the depression order from the console. These factors affect the transducer tilt but not the position of the tilt indicator.

a. Roll and pitch correction orders. These stabilization orders are incorrect if the azimuth beam is not centered on the same target as the QDA beam.

b. The alteration of depression order by the tangent solver to supply additional depression for targets on either side of the bow. The correction is necessitated by the non-variable fan-shaped beam.

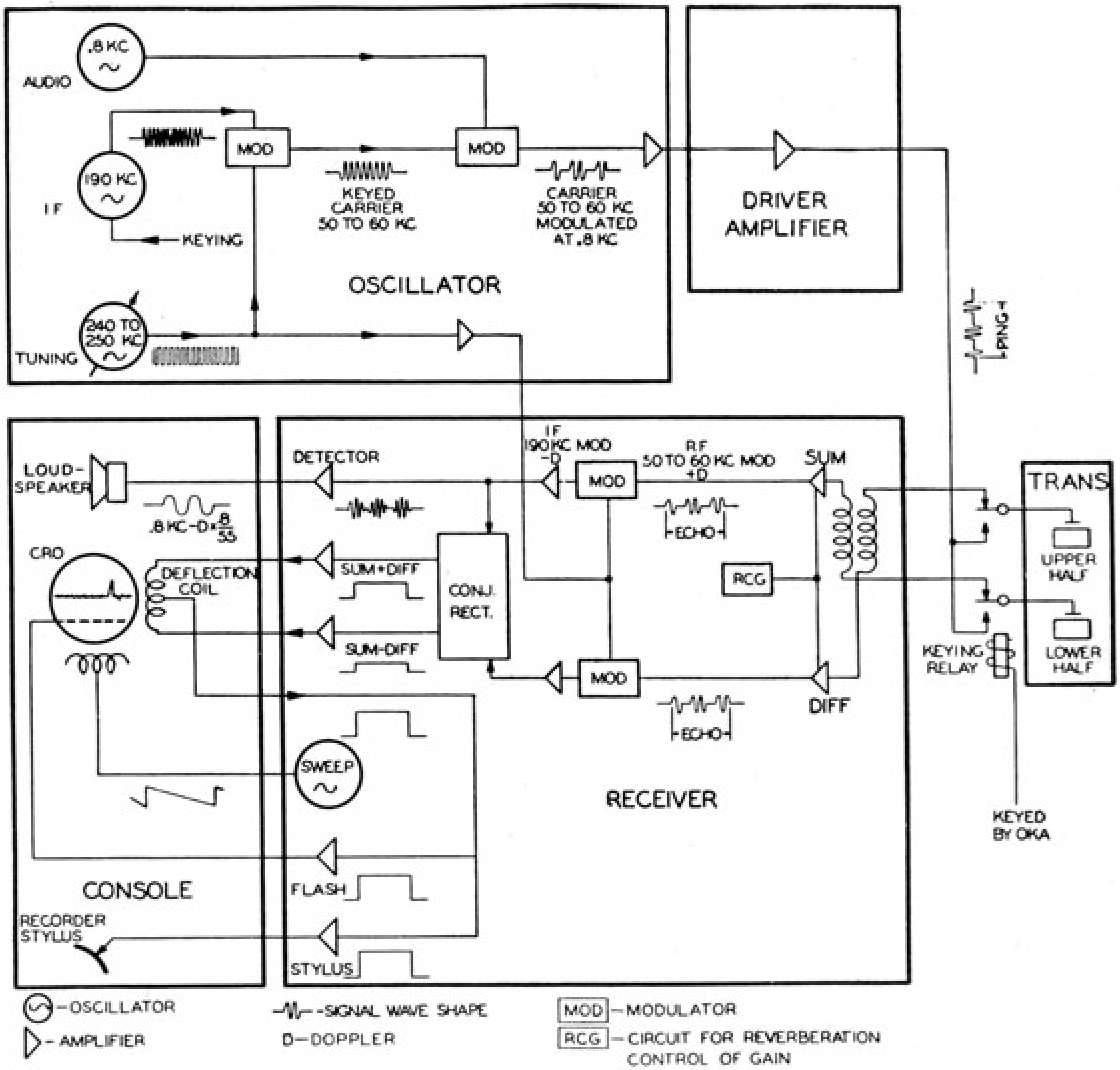


Figure 6-19 -Block diagram of the QDA transceiving system.

Transceiving System

The block diagram of the QDA transceiving system is shown in figure 6-19. The signal transmitted through the water is produced by inter-modulating the outputs of three oscillators shown in the figure. The 190-kc i-f oscillator is a fixed-frequency generator labeled "IF" because it establishes the intermediate frequency in the receiver. The unicontrol oscillator with a frequency range of from 240 to 250 kc (1) establishes the r-f carrier and (2) tunes the receiver to that frequency. The 0.8-kc audio oscillator originates the audio frequency that is heard when an echo or reverberation is received.

The oscillator with a frequency of from 240 to 250 kc is heterodyned with the 190-kc oscillator to produce a difference frequency of between 50 and 60 kc at the output of the first modulator. This difference frequency is the r-f carrier frequency. Because the 190-kc source is blocked except while the transmitter is keyed, the carrier exists only during the key-down condition. The carrier then is modulated with the audio frequency to produce the signal that is transmitted through the water. In the modulation process, negative half cycles of the relatively high level of the 0.8-kc oscillator blocks the modulator. The resulting modulated signal then consists alternately of a group of oscillations having a frequency of from 50 to 60 kc followed by an equal period of zero signal (figure 6-19). This modulated signal is composed of the carrier frequency and the side-band frequencies spaced at frequency intervals of 0.8 kc. The modulated carrier is amplified in the transmitter amplifier unit and is conducted through the transducer keying relays to the two halves of the transducer, which are connected in parallel.

The echo returning through the QDA transducer is similar to the outgoing signal except that the modulation. The Doppler always shifts this frequency to 190-D, the direction of shift being reversed by the heterodyning action.

In passing through the tuned i-f circuits, the higher-order side frequencies are filtered out and the modulated signal assumes a more nearly sinusoidal envelope. The detector rectifies this signal to produce an 800-cps signal similar to the envelope of the i-f signal. The frequency of the audio signal is 0.8 kc-D X(0.8/55); that is, the audio frequency is shifted by only about 1.5 percent of the original Doppler shift.

The reason for this shift can be understood if the modulated signal in the water is thought of as consisting of only three frequencies-the 60.000-kc carrier, the 59.200-kc lower side band, and the 60.800-kc upper side band. Because the Doppler effect shifts the frequencies by 0.7 cycles per kilo-cycle per knot of range rate, a range rate of 20 knots, closing, shifts the named frequencies to 60.840, 60.029, and 61.651 kc, respectively. The same frequency differences are preserved through the i-f circuit. The audio frequency produced when the signal is demodulated in the detector is the difference between the carrier and the side bands, or, in this example, 0.788 kc. Thus, the audio shift is only 12 cps (Doppler shift of the 0.8-kc modulation component), whereas the original Doppler shift is 840 cps. Obviously, the doppler would be very troublesome if the full shift of 840 cps were carried over into the audio signal, as it is in azimuth systems working in the frequency band of from 20 to 30 kc.

The virtual elimination of Doppler in the audio permits the use of a narrow pass filter in the audio circuit with significant benefits in noise reduction.

The signal in the *sum* channel, traced in the foregoing description, is formed vectorially by adding the signal outputs of the two halves of the transducer. The signal in the *diff* channel-the lower

frequency of the echo has been shifted by Doppler.

In the upper channel, marked "sum," the signals from the two halves of the transducer are combined to produce the signal that would be obtained if the transducer were not split. This signal is amplified in two r-f stages and passes through a modulator, where it is heterodyned against the frequency between 240 and 250 kc. In the absence of Doppler, the difference frequency is 190 kc-exactly the frequency of the i-f oscillator, although the signal shape is different because of the 0.8-kc

channel in figure 6-19-is formed from the vector difference between the signals in the two halves of the transducer. The *diff* signal is zero if the transducer beam is centered on the target in the vertical sense. The *diff* signal, which has the same character as the *sum* signal, is heterodyned in the same way as the *sum* signal. After suitable amplification, the *diff* signal, together with a portion of the *sum* signal, is fed into the conjugate

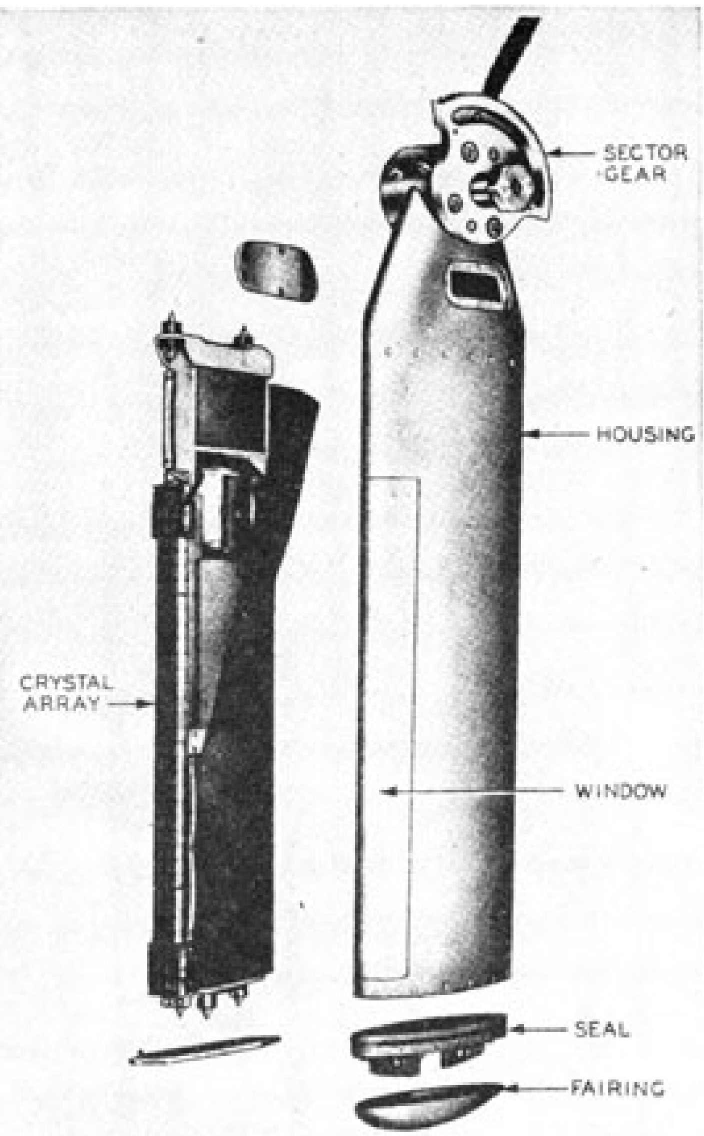


Figure 6-20 -QDA transducer.

tube to cause brightening when an echo is received, and (2) supplies current through the recorder-stylus chart-paper circuit to produce marks on the chart when echoes are received. These circuits are discussed more thoroughly in chapter 7.

Transducer

The transducer is of the ADP crystal type. It is shown in figure 6-20. The crystals are in an array approximately 20 inches long and 1 1/8 inches wide and are mounted in groups-each group ½ inch long and 5/16 inch wide-on a steel base-plate that has integral resonators directly behind the crystals. This arrangement provides efficient half-wavelength units with a nodal point at the face of the base plate. The array is connected electrically into separate halves about a horizontal centerline. Each half is connected to the high side of a transformer, the low sides of which provide impedance matches to the transmitter and receiver amplifier circuits. Gold-to-gold contacts between the crystals and their electrodes provide low interface electric resistance and consequently reduced heating. The gold surface is applied by evaporating gold onto the crystal faces and then bonding gold-plated metal foil to these

rectifier. As a result two d-c outputs are produced-one proportional to the *sum* signal plus the *diff* signal, and the other proportional to the sum signal minus the *diff* signal. These two d-c signals are transmitted to opposite ends of the vertical-deflection coil of the DDI oscilloscope in the console. If the *sum*-plus-*diff* signal is greater, the electron beam is deflected upward, indicating that the target is above the transducer beam. If the *sum*-minus-*diff* signal is greater, the spot is deflected downward, indicating that the target is below the beam. The two d-c signals are equal when the beam is on the target because the *diff* signal is then zero.

The two d-c signal-return currents, added together, flow out of the mid-point of the deflection coil, and this combined signal (1) supplies voltage to operate the intensity grid of the cathode-ray

faces.

The crystal array and resonator-plate assembly are mounted, with their long axis vertical, inside a streamline corrosion-resistant steel housing having a thin corrosion-resistant steel window that is almost sound-transparent. Also mounted inside the housing are two transformers, as well as a laminated baffle that attenuates extraneous signals through the back of the transducer. A blanket of the baffle material is assembled along the sides and back of the resonator assembly and helps to reduce the effects of reflections within the housing. At the top of the housing is an integral pivot block, on which are mounted a sector gear and trunnion bolts for attachment to the hoist-tilt mechanism. A cover plate near the top of the housing provides access to a cable seal and filling plug. Another cover at the bottom of the housing provides access to a second filling plug in a rubber-gasketed plate and ring assembly, which seals the housing.

The transducer is vacuum-filled with approximately 1 gallon of electrical-grade castor oil, from which the air and water vapor have been removed.

The transducer is a highly efficient reciprocal converter of electric into acoustic energy over a frequency range of 50 to 60 kc. This wide range

makes possible a choice of frequency that permits simultaneous operation of equipment by several ships in the same area. The beam pattern is very sharp in the vertical plane so as to permit the accurate determination of target depth. In the horizontal plane the pattern is broad so that contact with the target can be maintained over a wide angular range on either side of the bow of the ship. Figures 6-21 and 6-22 illustrate typical beam patterns in both planes.

The crystal array of the transducer is divided electrically into top and bottom halves for obtaining depression deviation indications.

Transmitter

From the output of the second modulator a voltage of from 50 to 60 kc ±0.8 kc is delivered to the grid of the transmitter preamplifier, a type 6SJ7 tube. From the preamplifier output this voltage is used to excite the control grids of the transmitter-amplifier stage, which consists of six type-807W beam-power tetrodes connected in push-pull with three tubes in parallel on each side. The output of the transmitter is connected to the transducer through a tuning retard coil, which has several taps that facilitate tuning out the

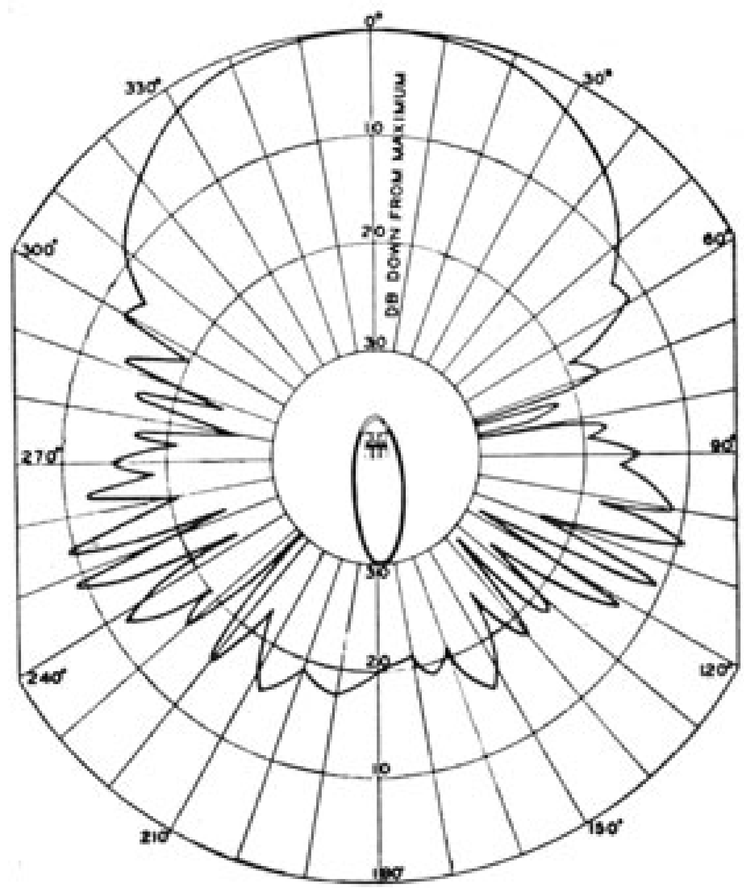


Figure 6-22 -Typical horizontal beam pattern of a ODA transducer at 55 kc.

capacitance of the transducer and approximating an impedance match between the amplifier and the transducer. The plate-to-plate impedance of the output circuit should be 3,300 ohms for maximum power output, which is 150 watts.

Summary

The QDA target depth-determining equipment and the azimuth sonar equipment when operating together provide complete and continuous information regarding the location of a submerged object. The sole function of the QDA is to determine the depth of a submerged object, whereas the azimuth equipment determines the range and bearing of the target.

The QDA determines target depth indirectly because it is an echo-ranging equipment that measures the time lapse between transmission and echo. The

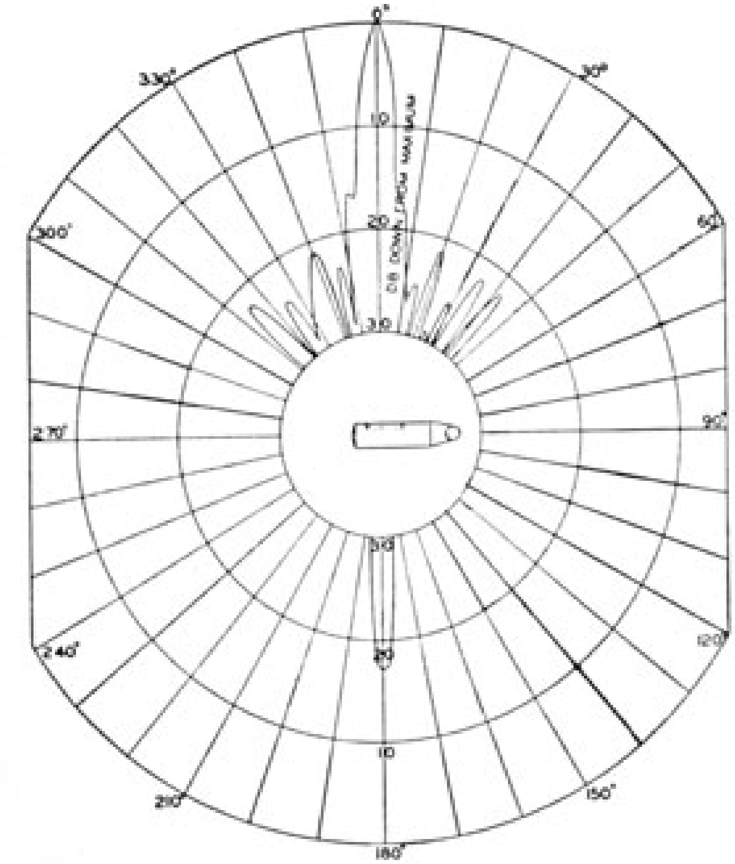


Figure 6-21. -Typical vertical beam pattern of a QDA transducer at 55 kc.

vertical velocity of the sound beam is measured by a chemical recorder that has a variable-speed stylus. The stylus speed is a function of the sine of the depression angle times the velocity of sound. The stylus speed is controlled by an associated equipment-the OKA-1 recording-resolving equipment-but the basis of the

stylus speed is the QDA beam-depression angle and the velocity of sound in water. To obtain accurate depth solutions the equipment has a cathode-ray tube used as a depth-deviation indicator.

Because the target-depth information is employed at various locations on the ship, a means for transmitting the information is provided by an optical cursor that positions a synchro transmission system.

While the sonar vessel is closing a target, the depression angle increases-slowly at first, and more rapidly as the range is decreased. To relieve the operator of most of the burden of following the target, the OKA-1 provides aided tracking by (1) computing the depression angle theoretically required-known as the computed target

If the beam is bent downward, the stylus speed is increased for a corresponding period because the depression angle is greater than the ordered angle. Layer-depth contacts, closed by the stylus carriage, are adjustable and bring about the transition from the mixed-layer travel rate to the refraction-layer rate. The layer-depth contacts in the depth recorder and the two sound-velocity controls in the OKA-1 resolver are set according to the information from bathythermograph readings.

SONAR INSTALLATIONS

The sonar system installed on antisubmarine vessels is composed of several sonar and fire control equipments operating in a reciprocal-information and control network. The purpose of the system is to fix a submarine's position *once contact has been established* and to solve the necessary fire control

depression, *cEtq*-and (2) transmitting this synchro order to the QDA tilt-order system. The QDA operator then adds corrections, *jEtq*, to this order, guided by the indications of the DDI.

The QDA beam must be stabilized against roll and pitch, and the depression orders originated by the QDA operator must be modified accordingly. This action is accomplished by the stabilization computer, with which a stable element is associated.

The transducer cannot be trained in azimuth. When the target is dead ahead, the target depression corresponds to the transducer tilt-if the sound path in the water is a straight line. When the target bears off the bow, contact still can be maintained because the transducer beam is very broad in an athwartships direction, but a greater tilt is required to keep the fan-shaped beam on the target. This correction to the depression angle is made by the tangent solver, which is a unit of the stabilization computer.

The sound path through the water generally undergoes some bending, principally because of unequal temperatures at various depths. The OKA-1 equipment makes the correction for bending by varying the stylus speed of the depth recorder. While the sound energy is passing through the mixed layer, the path does not bend and the stylus speed is constant. During this period the stylus speed is based on the sound velocity near the surface and the ordered depression angle. Below the mixed layer the sound energy travels at a speed that varies with depth, and as a result the beam is bent.

problems to assure a kill. The system is an attack system and not a search system. There has always been a need for a fire control system in antisubmarine warfare, but the need did not become acute until the advent of the high-speed, deep-diving submarine. At the outbreak of World War II the depth of the submarine was approximated by the conning officer from the range at which contact was lost because of the sound beam passing over the U-boat. By the time the lost-contact range was reached the anti-submarine vessel was on its attack course, and was already starting to lay the depth-charge pattern.

In spite of this difficulty, the method was fairly effective against the old type of submarine with riveted construction because the pressure hull could be ruptured by near-misses. With the advent of the modern welded-hull construction, which can stand terrific pressures and stresses, it became necessary to score a direct hit on the submarine to do a reasonable amount of damage. The underwater fire control system furnishes the precise information necessary to score these direct hits. Furthermore, it supplies the information until a very late stage in the attack. The older single-sonar search system frequently lost contact at ranges up to 600 yards. As a result, the submarine had ample time to take evasive action which could not have been detected by the attacking vessel.

A typical installation aboard an antisubmarine vessel may consist of a QGB or QHB, a QDA, an OKA-1, a Mark 4 director, and a stabilization computer with its associated stable element.

Previous Chapter Sonar Home Page Next Chapter

CHAPTER 3

SOUND RECEPTION AND DETECTION BY LISTENING

Introduction

Sound waves can be received only if a device that will absorb a fraction of the incident energy and convert it into a detectable form is placed in their path. Such a device is called a *receiver*. The proper type of receiver for a particular application depends upon (1) the frequency, amplitude, and form of the sound wave; (2) the type of transmitting medium; and (3) the ultimate object for which the sound energy is required.

A *resonant receiver* is designed to operate with maximum efficiency at some particular frequency. A nonresonant receiver is designed for use when a reasonably uniform response is desired over a given range of frequencies. If the primary concern is faithfulness in the reproduction of waveform, a nonresonant receiver is required. However, if it is necessary to receive sound waves of a particular frequency to the exclusion of other frequencies that may be present in the medium, a resonant receiver is required.

Most sound receivers function to transform the mechanical energy that they absorb directly or indirectly into electric energy. The electric energy representing the sound signals may be portrayed visually, or the sound signals themselves may be reproduced as sound energy by a loudspeaker.

Human Ear

Sonar equipment that presents sound signals by means of a loudspeaker is useless unless there is an operator to hear and interpret the sound waves radiated to the surrounding air. The capabilities and limitations of the operator, whose task it is to interpret the sounds issuing from the listening gear, are important in determining the success or failure of its mission. For this reason, the following discussion on the physics and psychology of hearing is included, even though it is not strictly a part of the theory of underwater sound. Note that

Most of the lengthy arguments that were expounded on this question could have been avoided had there been adequate theories of sound and hearing. Today *sound* means waves, which travel in the air, water, or other medium. Thus the answer to the crashing-tree question is yes. Sound is to be distinguished from the sensation of hearing, or auditory sensation, which is a phenomenon occurring in a human being or animal. There was no auditory sensation in the crashing-tree example. To clarify the distinction between a sound and the sensation produced by a sound, the sound is often called the *stimulus*. Ultrasonic waves are sound, but they do not stimulate the sensation of hearing in human beings; they are thus not a stimulus of auditory sensation.

this discussion deals primarily with airborne sound.

Confusion sometimes arises between the objective physical phenomenon of sound and its subjective perception by a listener. The reader is doubtless familiar with a philosophical problem that agitated the ancients, which was formulated somewhat as follows: A tree crashes in a forest, and no living being is present to perceive the fact. Is there any sound?

THEORY OF HEARING

In this study it is not essential that a physiological study of the ear be made. Of particular interest here is that part of the inner ear called the

---

cochlea which has a major part in the hearing process. It is a spiral tube, divided into galleries by a longitudinal membrane-the *basilar membrane*, which is a sort of carpet of nerve endings. The nerve endings of the basilar membrane are transverse fibers that vary systematically in length. The short fibers respond to sound waves of high frequencies; the long fibers respond to sound waves of low frequencies. That is, the position of the point of maximum stimulation depends on the frequency of the tone.

In response to a complex sound, the basilar membrane vibrates with a certain pattern, perhaps having several maxima, depending on the frequency components in the stimulus. The auditory nerve endings are distributed along the basilar membrane in such a way that they can transmit this pattern to the brain, which interprets it in terms of the pitch, loudness, and quality of the sound. The location of the vibration pattern on the basilar membrane determines the pitch

The number of responses of a given nerve fiber depends on the strength of the stimulus; moreover the number of nerve fibers excited increases with the intensity of the stimulus because (1) a greater area of the basilar membrane is activated and thus the stimulus pattern on the membrane takes in nerve endings over a wider area, and (2) the high intensity excites nerve fibers having higher normal thresholds of stimulation. It seems reasonable, therefore, to correlate the sensation of loudness with the total number of nerve impulses arriving at the brain.

NUMERICAL DATA CONCERNING THE EAR

The preceding theory of hearing suggests how the structure of the ear enables it to respond to frequency and intensity characteristics of a sound. Although it is a theory that has not been verified in all details and is subject to revision, it should help in understanding some of the pages which follow. However, the following facts are independent of the correctness of this theory.

Frequencies of from 20 to 20,000 cycles per second can be heard by a normal, young ear. A change in frequency of less than one-half of 1 percent results in a perceptible change in the pitch of pure tone. This phenomenon takes place at 10,000 cycles per second only if the listening level is comfortably loud. As the duration of the tone signal becomes shorter, the ability to hear pitch changes decreases. This relation is shown

sensation. Loudness is associated with the magnitude of the vibration.

The relation between the perceived loudness of a sound and the magnitude of the stimulus on the basilar membrane is explained as follows: The auditory nerve contains about 3,000 nerve fibers which, analogous to a telephone cable, connect the cochlea to the brain. Each nerve fiber responds according to the "all-or-none" law; that is, when it is stimulated sufficiently to respond at all, it responds at full strength. The response of a nerve fiber is analogous to the discharge of a condenser. The strength of the discharge is independent of the intensity of the sound, but the number of discharges per second does depend on the magnitude of the stimulus in the following manner.

The discharge of a given nerve fiber is followed by a "refractory period" during which the nerve cannot react. This period is about 0.001 second; thus no single nerve fiber can respond at a rate greater than about 1,000 times per second. The refractory period is followed by a "relative refractory period" of about 0.003 second during which the nerve gradually recovers its sensitivity. Thus a very weak tone of, say, 1,000 cycles per second may cause a given nerve fiber to discharge no more rapidly than about 300 times per second, whereas with an intense tone of that frequency the nerve may respond up to 900 times per second.

in figure 3-1, where the least-perceptible frequency change plotted against the signal duration. It is interesting to note that at 1,024 cycles per second

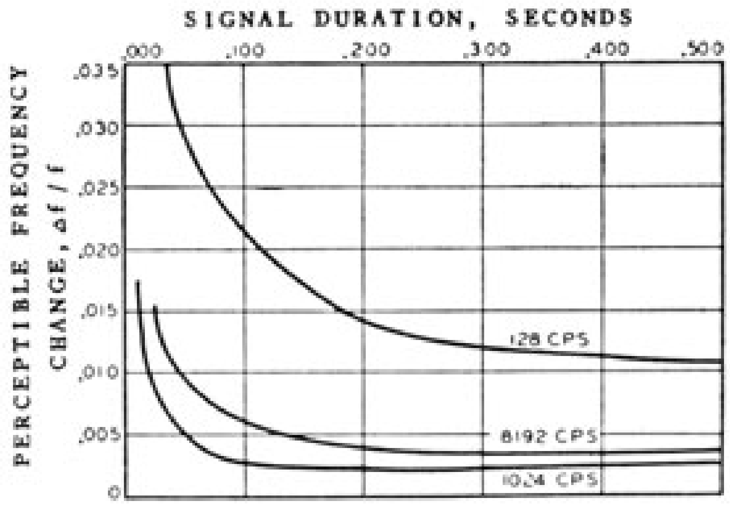


Figure 3-1. -Threshold of frequency discrimination for several frequencies as a function of signal duration.

the length of the signal affects pitch discrimination only if the signal length is less than 0.1 second. This fact is important in doppler discrimination in echo ranging.

The ear is most sensitive at frequencies between 1,000 and 5,000 cycles per second, where a sound intensity of approximately 10^{-16} watt/cm^2 can be heard. A sound intensity of approximately 10^{-4} watt/cm^2 produces a sensation of pain rather than of hearing. Thus the ear has a dynamic range of about 120 db at frequencies around 1,000 cycles per second.

A rapid change of 1 db, or slightly less, in the level of a pure tone can ordinarily be perceived at all frequencies between 50 and 10,000 cycles per second if the listening level is comfortably loud.

The ability to detect changes in level is less for randomly fluctuating sounds, such as noise, than for pure tones. However, a simple rhythmic variation is very easily perceived, particularly if it is cyclic at the rate of about 3 per second.

The ear requires approximately 0.2 second for the sensation of loudness to catch up with a sudden increase or decrease of sound level. These dynamic properties seem to be determined by neural rather than mechanical processes. They influence the response of the ear to tones of short duration such as those used in echo ranging.

Sounds having the same pitch and loudness may produce different sensations if their spectra are different. The oscilloscope fails to operate properly before the sound intensity has reached zero. This minimum intensity to which the oscilloscope responds depends on two factors. One is the amount of energy dissipated in the various parts of the microphone; the other is the self-noise of the oscilloscope, the microphone, and the circuit. The oscilloscope will not operate properly unless the signal is at least as intense as the self-noise. The minimum sound level that will cause the device to operate properly is its threshold.

Suppose that the receiver is now replaced by a human ear, and the same procedure is followed. A precisely analogous situation results, and for much the same reasons. The ear receives the sound energy incident on it, is stimulated mechanically, and the mechanical energy then is converted into some form of nerve energy which activates the brain. Some of the incident energy is dissipated in this process. Corresponding to the self-noise of the receiver, there are sounds generated by breathing and by the circulation of the blood. Thus there is a minimum level which must be exceeded by a sound before it can be heard. This threshold of hearing corresponds to the threshold of the microphone-oscilloscope system.

The value of the threshold of hearing differs among people. We say that their acuity is different. The average value of the threshold of hearing also depends on the frequency. At 64 cycles per second the pressure of the threshold of hearing is 0.12 dyne/cm^2; it decreases more or less uniformly with increasing frequency up to about 3,000 cycles per second, at which frequency the pressure is 0.000041 dyne/cm^2. This value corresponds to the lowest limit of sensitivity mentioned earlier. Above 5,000 cycles per second it increases with frequency until at 18,000 cycles per second it is 4.1 dynes/cm^2.

MASKING

Under all ordinary circumstances, we hear many sounds at once but are usually able to concentrate on the wanted sounds and ignore the unwanted background. This background is always present. Even in a very quiet place the self-noise produced by the normal internal processes of the human body

The general term "quality" is used to describe the difference in the complex sensations they stimulate. These differences may be sufficient to influence the masking of one sound by another. Because masking is a primary factor in preventing the detection of signals, its general principles will be discussed in greater detail than has been accorded to the other aspects of hearing.

THRESHOLD OF HEARING

The *threshold of hearing* may be illustrated by the following experiment. A microphone is placed near a sound source which produces a pure tone of controllable intensity. Apart from this sound the experimental location is to be very quiet. The microphone converts the mechanical energy of the sound into electric energy which can be used to operate some device, such as an oscilloscope.

Beginning with a sound intensity of moderate value, the intensity of the tone is gradually reduced.

becomes audible. Thus there is complete analogy between the ear and an electronic receiver of sound. This analogy is

close enough to permit the frequent use of the word "receiver" with reference to the ear as well as to electronic devices.

Although unwanted sounds can be ignored to a considerable extent, their presence does interfere with the ear's ability to detect another sound. This interference is called *masking*. Masking is the increase of threshold level caused by the unwanted sound.

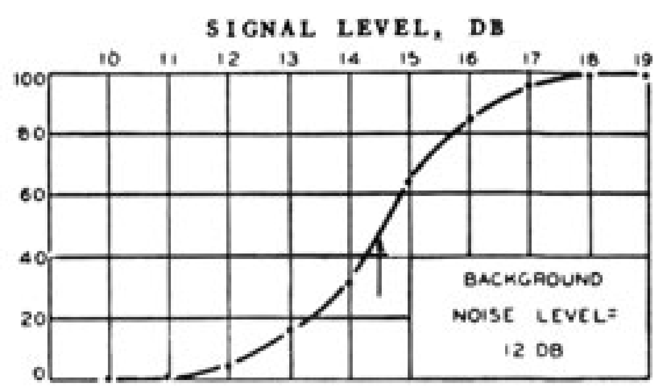


Figure 3-2 -Probability of recognition of a pure tone in a background of a noise at a constant level of 12 db.

The level at which a particular sound becomes audible differs from the threshold of hearing by an amount depending on the extent to which the background noise masks the signal. This level is the *masked threshold*; it is the level of the signal when it is audible above a particular background noise 50 percent of the time. The masked threshold therefore applies to the signal-noise pair, not to the signal alone, although it is measured by the level of the signal alone. The value of the masked threshold is, however, determined by the level of the noise. Raising the level of the noise raises the masked threshold of the signal.

The variable acuity of a listener introduces the need for the phrase "50 percent of the time." Not only does the threshold of a signal under identical conditions vary from individual to individual, but the same individual sometimes hears a signal and sometimes not, even though the levels of signal and masking noise are the same on the various occasions.

This problem may be clarified by describing a typical experiment designed to measure the masked threshold. Arrangements are made so that a number of listeners will hear the background noise at a constant and known level. Other arrangements are made for producing a series of signals at various levels. Care is taken so that the listeners cannot determine when or at what level a signal is produced except by hearing it; they receive no cues from the person administering the test nor from each other. The administrator

Note that, there is no abrupt transition from inaudibility to audibility. Instead, the probability of hearing the signal increases gradually from zero to 100 percent over a 5-db range of levels. This complication was not considered in discussing threshold levels in the preceding pages. Fundamentally, there is no one level at which the signal is "just audible." To avoid confusion, threshold levels are usually defined as the level at which the recognition probability is 50 percent; but, when necessary, other percentages may be used, provided they are specifically indicated. Figure 3-2 shows that the 50-percent masked threshold is 14.5 db, the 90-percent threshold is 16.4 db, and the 10-percent threshold is 12.6 db.

This difference between the threshold level of the signal and the level of the background is called the *recognition differential*. In the example the recognition differential for 50-percent recognition is thus 2.5 db (14.5-12.0).

PSYCHOLOGICAL CHARACTERISTICS OF SOUND

How does the ear distinguish between a specific sound and all the other sounds that form a background for it? Everyday experience suggests the answer. A boatswain shouting orders must rely chiefly on his ability to produce sounds of an intensity great enough to override the clamor of winches and other noises. A shrill whistle produces a sound that is audible, even though the intensity of the background is incomparably greater than that of the whistle. In this case the perception is due partly to the pitch difference between the signal and the background noise, and partly to a decided difference in the quality of the two sounds. A rhythmic drumbeat is audible over many noises. Before the days of telephone and radio the common method of transmitting

records the level of each signal and, after a suitable interval, instructs each listener to vote yes or no as to whether he heard the signal.

Each level of the signal is presented 5 times to 10 listeners, so that the total number of votes for each level is 50. The recognition probability is the percentage of yes votes for a given level. This probability is plotted as a function of level in figure 3-2.

---

orders to masses of troops was to use drumbeats of various rhythmic patterns. Bugle calls with very decided rhythm utilized the advantages of all the factors mentioned.

To sum up, the sensations produced by sound have at least four distinctive characteristics: (1) Loudness, (2) pitch, (3) quality, and (4) time pattern. In the recognition of a particular sound, all four of these characteristics probably contribute to differentiate it from others heard simultaneously. In experiments, however, the effect of each characteristic can be isolated.

Loudness, *pitch*, and *quality* are psychological, rather than purely physical, terms. That is, they directly characterize the sensation and only indirectly the sound. It is customary to say loosely that loudness is determined by the level of a sound, pitch by the dominant frequency, and quality by the spectrum. This explanation is over-simplified. A more careful examination discloses that in determining any one of factor that determines loudness, we may inquire as to the mathematical relation between intensity and loudness. It appears that this relation is not a simple proportionality-that is, when one sound is said by most people to be "twice as loud" as another, the intensity of the one is not twice, but approximately 100 times, the intensity of the other. In general, loudness is more nearly proportional to the level of the sound in decibels. A barely perceptible increase of loudness usually accompanies a sudden increase of 1 db in sound level, whether the original level was 5 or 50 db.

Another characteristic that can be used to differentiate sounds is their direction of arrival. In simple cases, this direction coincides with the direction of the source from the listener. The *binaural effect* is the ability of a human with two ears to determine the direction of a sound source. This sense of sound direction depends primarily on the difference in phase (or time) of the waves reaching the two ears, although it depends partly on the difference in intensity of the sound received in the two ears. The binaural effect is similar in principle to the split transducer used with *bearing deviation indicators* (BDI).

In the early days of sonar, attempts were made to use the binaural effect to determine the direction of underwater sound. These listening devices used two receivers placed along a baseline varying from several feet to several hundred feet. This procedure virtually increased the baseline between the two ears.

the three, all the physical characteristics of the sound play a part. Loudness, it is true, is determined primarily by the level of the sound, but it is influenced also by the frequency and spectrum. It has been demonstrated experimentally that a moderately high frequency is perceived as being louder than a low frequency of the same intensity. This fact is almost implicit in the discussion of the threshold of hearing given above. If the frequency exceeds about 14 kc the reverse is true, and ultrasonic sound of any level is inaudible. Pitch, in its turn, is determined largely by the dominant frequency of the sound waves but is influenced also by the level and the other characteristics of the spectrum. Quality is principally a matter of spectral distribution; and the time pattern may consist of systematic changes in any of the other three psychological characteristics.

One point is worthy of particular emphasis. Ignoring the fact that intensity is not the only

An early device of this type, designed for underwater listening, consisted of two hollow rubber spheres mounted on the ends of a pipe about 4 feet long. Projected through the hull of the ship, the receivers were separately connected over lines of equal length to the two ears. The tube might then be turned until the sound appeared centered in the head; at which time it should be on a line perpendicular to the baseline of the receivers.

Doppler Effect

DEFINITION

The Doppler principle applicable to all wave motion was developed by the Austrian physicist, Christian Doppler (1803-1853). This principle shows that when there is a relative motion between the source of a wave motion and a receiver the apparent frequency at the receiver differs from the frequency at the source. The Doppler principle has important operational applications in sonar.

If an observer is moving *toward* a source of sound, he hears a tone the pitch of which is *higher* than when he is at rest. If the observer is moving

away from the source of sound, he hears a tone the pitch of which is *lower* than when he is at rest.

Thus the frequency of the sound appears to increase when a observer moves toward a source and appears to decrease when he moves away from it. Similarly, if the source is moving toward the observer, the frequency is *higher*; if the source is moving away from the observer, it is *lower*.

The apparent frequency of the sound is found as follows: When the observer is at rest, the number of waves he receives each second is F_o, the true frequency of the sound. When the observer moves toward the source, he receives more sound waves in each second than when he is at rest. If his mean range rate is *dR* (in feet per second), the additional number of waves received per second are those that occupy the distance by which the range is changed in 1 second. Because the distance between successive waves is the wavelength λ, this number is dR / λ.

If the relation for the velocity *v* of the sound,

$$v = F_o\lambda, \quad (3\text{-}1)$$

is used, the number of additional waves received is $F_o dR / v$. The apparent frequency, *F*, is the total number of waves received each second and is therefore given by

$$F = F_o(1 + dR/v). \quad (3\text{-}2)$$

this reduction is subtractive and not proportional-that is, the receiver subtracts a constant amount, F_H, from the received frequency, F_E, so that the audio frequency of the output is

$$f_E = F_E - F_H \quad (3\text{-}5)$$

If this equation is applied to equation (3-4) the audio frequency of the echo is

$$f_E = F_o - F_H \pm (2F_o dR)/v, \quad (3\text{-}6)$$

or

$$f_E = f_o \pm (2F_o dR)/v, \quad (3\text{-}7)$$

Here $f_o = F_o - F_H$ is the audio frequency of the echo for a zero range rate. The difference $f_E - f_o$; that is, the quantity $\pm 2F_o dR/v$; is called the *absolute doppler shift*. It is proportional to F_o, and independent of F_H and f_o. This fact is very important because the transmitted frequency, F_o, is much greater than the heterodyned audio frequency f_o. Because the Doppler effect is to shift the frequency by 0.7 cycle per kilocycle per knot of range rate, if *dR* is expressed in knots and F_o, in kilocycles, the doppler shift is

$$f_E - f_o = 0.7F_o dR \text{ cps (approx).} \quad (3\text{-}8)$$

If F_o, is 24 kilocycles,

$$f_E - f_o = 17dR \text{ cps (approx).} \quad (3\text{-}9)$$

This shift can be very appreciable. If the sonar ship and the target are on opposite courses, and one is moving at 25 knots and the other at 20, the shift is 45 X 17=765 cycles per second, and a band pass of twice this quantity, or 1,530 cycles per second is required. Because f_o, is commonly 800 cycles per

When the observer is in motion away from the source, the plus is replaced by a minus-

$F = F_o(1 - dR/v)$. (3-3)

If the source is receiving echoes from a target, the Doppler effect occurs twice, so that the frequency of the echo F_E, received at the source is

$F_E = F_o(1 \pm 2dR/v)$. (3-4)

Equation (3-4) gives the apparent frequency of the echo when the range rate is dR; the positive sign is used if the receiver and the source are moving toward each other, the negative if they are moving away from each other.

The equations apply to the ultrasonic frequency of the sound in the water. To make this sound audible, the received waves are heterodyned in the receiver. This heterodyne receiver reduces the frequency by a constant amount. Note that

second, this frequency shift is important in determining the width of the band pass of the sonar receiver. Circuits may be used to eliminate this shift when it exceeds the band pass of the receiver. One such circuit is called *own doppler nullifier* and the other, *target doppler nullifier*. These circuits will be discussed later.

APPLICATION TO ECHO RANGING

In echo ranging the operator does not hear the outgoing ping, because the equipment is on send and the receiver is blocked. Therefore, he cannot compare the frequency of the returning echo with that of the outgoing ping. However, he can compare the frequency of the echo with that of the

reverberation heard immediately after the ping is emitted. This comparison has an important effect. The difference between the reverberation and echo frequency depends only on the target's absolute motion through the water and its direction relative to the sound beam. It is independent of own ship's motion.

For example, suppose a ship is moving with its sound beam directed dead ahead and with a velocity, V_1, which is also the range rate, *dR*, if the echo is from stationary objects (scatterers). Just as with an echo from a moving target, the relative motion between the source and the scatterers causes the reverberation frequency to increase. From equation (3-7), the reverberation frequency after heterodyning is

$f_R = F_o + (2F_oV_1)/v$ (3-10)

If a submarine is approaching the echo-ranging ship with a speed V2, the relative speed or range rate is

$V = dR = V_1 + V_2$ (3-11)

and from equation (3-7), the audio frequency of the echo is

$f_E = f_o + (2F_oV_1)/v + (2F_oV_2)/v$ cps. (3-12)

A comparison of equations (3-10) and (3-12) shows that the audio frequency of the echo exceeds that of the reverberation by an expression that does not contain V_1, the speed of the sonar vessel.

thus, for an approaching 20-knot submarine, the frequency of the echo is 340 cps above the reverberation frequency. The quantity Δf is known as the target doppler. Because operationally it is much more important than the absolute doppler shift, it is frequently called simply *Doppler*. It is "up-doppler" if the submarine is moving toward the echo-ranging ship and "down-doppler" if it is moving away from the echo-ranging ship. Another useful characteristic of target doppler is that it is proportional to the speed of the target. Hence it can give information concerning the motion of the target. A trained operator can estimate also the probable aspect of the target with considerable accuracy from the change in target doppler.

In the foregoing example, it s assumed that the course of the target is directly toward (or away from) the echo-ranging gear. t may be shown that, in general, V_2 is not the actual speed of the target, but is its range rate relative to a stationary point, *P*. This point, *P*, momentarily coincides with the sonar projector but must be considered stationary even though the sonar is moving.

The importance of target doppler in echo ranging is immediately evident. It is a common experience that a difference in pitch between two tones is a great aid in hearing them; and even a very weak tone can often be distinguished from others if its pitch differs markedly. Thus target doppler is a great aid in detecting echoes against a reverberation background but not against noise. The ability of the operator to estimate the difference in frequency between reverberation and echo depends on the ping length.

Many "false" echoes are received from floating debris, kelp, and unknown causes. These echoes do not show the effect of target doppler. Thus a final important application of target doppler is in the identification problem.

If F_o, is 24 kilocycles,

$\Delta f = 17V_2$ cps; (3-14)

Ear in Underwater Detection

LISTENING

Detection of underwater objects by *listening* for the sounds they emit is known as listening. Sounds made in the sea are easily detected by the use of listening equipment. Listening, the oldest method of detection, was used in World War I in a very crude, but nevertheless effective, form. The success of detection by listening is primarily dependent on the ability of the operator to hear and properly evaluate these sounds delivered by the listening equipment.

Any listening system must consist of (1) a hydrophone, (2) an electronic receiver, (3) a bearing indicator, and (4) a speaker or headphones. The sound-listening problem for the operator consists primarily of learning to distinguish between (1)

---

sounds emitted by another ship's machinery through the hull and from the propeller and (2) the multitude of other sounds that exist in the ocean.

There is always the problem of background noise, which may make the sounds to be detected unrecognizable. As pointed out earlier, the characteristics of the ear enter into this problem. During World War II many persons were found to have hearing that was defective for sonar work.

Echo ranging and the listening problem differ materially in several ways.

In echo ranging, the searching vessel projects a sound signal into the water intentionally with the expectation that

Listening is used chiefly by submarines. A surface vessel produces considerable noise, and this noise interferes with the detection of the sounds of other ships-especially the low sounds of submarines. On the other hand, this difference in the noise output enables a submarine to detect the presence of a surface vessel rather easily. An anti-submarine vessel, moreover, will generally not use evasive tactics. Therefore it will not hesitate to emit a powerful signal into the water, and thus gain the advantages of echo ranging; whereas a submarine will hesitate to reveal its presence by echo ranging except in the last stages of an attack.

In order for listening to be a tactical aid, the sound operator by use of his ear must be able:

1. To distinguish the sound emitted by the target from the usual background noise.
2. To distinguish between the various kinds of ship sounds with a view to possible identification of the type of vessel emitting them and to obtaining information on the ship's

the sound will strike a target and that enough of the energy will be returned by the target to the transducer to activate the receiver so that the operator can recognize the echo. The primary source of the sound is in the searching vessel; the target is only a secondary source. The transmission of the sound is a two-way process. In listening, on the other hand, the sound signal is emitted by the target itself, which therefore is the primary source. Listening is hence a one-way process.

This fact suggests that losses by transmission should be smaller in the case of listening, and that detection should be possible at longer ranges by listening than by echo ranging, provided that the sound output of the target is comparable to that of the standard echo-ranging transducer. however, the noise output of most targets is less than the output of a standard transducer. Even the noisiest type of ship, a large battleship moving at high speed, has an over-all output of sound of about the same level as a standard transducer. Furthermore the sound from a transducer is a pure tone, because the echo has frequencies that are restricted to a band of about 200 cycles. On the other hand, the sound from a battleship has components of a wide range of frequencies, and hence is more easily masked by the background noise.

Nevertheless, conditions are frequently such that ships are detected by listening at ranges of 10,000 yards and more, whereas echo ranging is rarely effective beyond 3,000 yards. Echo ranging enables the range and bearing of the operating conditions.

3. After detecting and perhaps partially identifying a target, to obtain information concerning its approximate location and motion while it is still at comparatively long range.

These considerations suggest the value and purpose of the investigation of ship and submarine sounds. Such information will aid in the problem of the control or possible elimination of revealing noises. The basic principle in this problem is the same as that underlying visual camouflage-to render the target inconspicuous by making it resemble its background. Thus the sounds that are unintentionally and unavoidably emitted should, in the ideal case, have spectra that are very similar to that of the background noise.

Another application is in the design and operation of acoustic mines and in the prediction of their actuating ranges. This application, as well as the defense against mines of this type, requires a knowledge of the sound emitted by the vessels against which they are to be used.

BACKGROUND NOISE

There are two principal sources of background noise-*airborne noise* and *amplified noise*. When using listening equipment, the operator depends almost entirely on his ears, unaided by any form

target to be determined accurately; listening gives the bearing quite accurately, but provides little or no information on the range except in specialized equipment.

---

of recorder or other apparatus. Occasionally a decibel meter or "magic eye" is available for supplementary quantitative information. His task is reduced to detecting and recognizing a wanted signal against the background of all the other sounds that impinge on his ear. These sounds are many and complex.

In the discrimination process, the operator distinguishes between wanted sounds of the signal from the target, and the unwanted sounds that are picked up or generated by the receiver as well as airborne sounds from his surroundings.

Airborne sounds often may be a limiting factor. Listening in an airplane for the signals from a sonobuoy sometimes is limited by this type of noise, which often is referred to as "local noise" or "room noise." The signal can be made more perceptible by increasing the amplification of the receiver; for in this case the airborne noise is not amplified and the signal-to-noise ratio is increased.

The desired signal is but one of the many sounds that are amplified and heard by the operator.

These sounds originate in the sea and in the listening vessel itself, and they constitute a masking background for the signal. Increasing the gain of the receiver in this case does not help, for the background noise also is amplified with the signal. Noises that are created in the receiver itself also are amplified and, mask the desired signal, the same as those sounds that are picked up by the hydrophone.

The sources of the circuit noise are (1) thermal agitation of electrons in the tuned input circuit, (2) tube noise, (3) hum due to man-made disturbances, and (4) vibration of tube elements resulting in "microphonics."

Figure 3-3 shows the complete classification of background noise. This figure shows that self-noises are (1) circuit noise, (2) hydrophone motion, and (3) noise from own ship such as vibration and turbulence caused by the ship's motion.

The other important sources of background noise are classified as *ambient noise*. Ambient noises are (1) sea noise, due principally to the

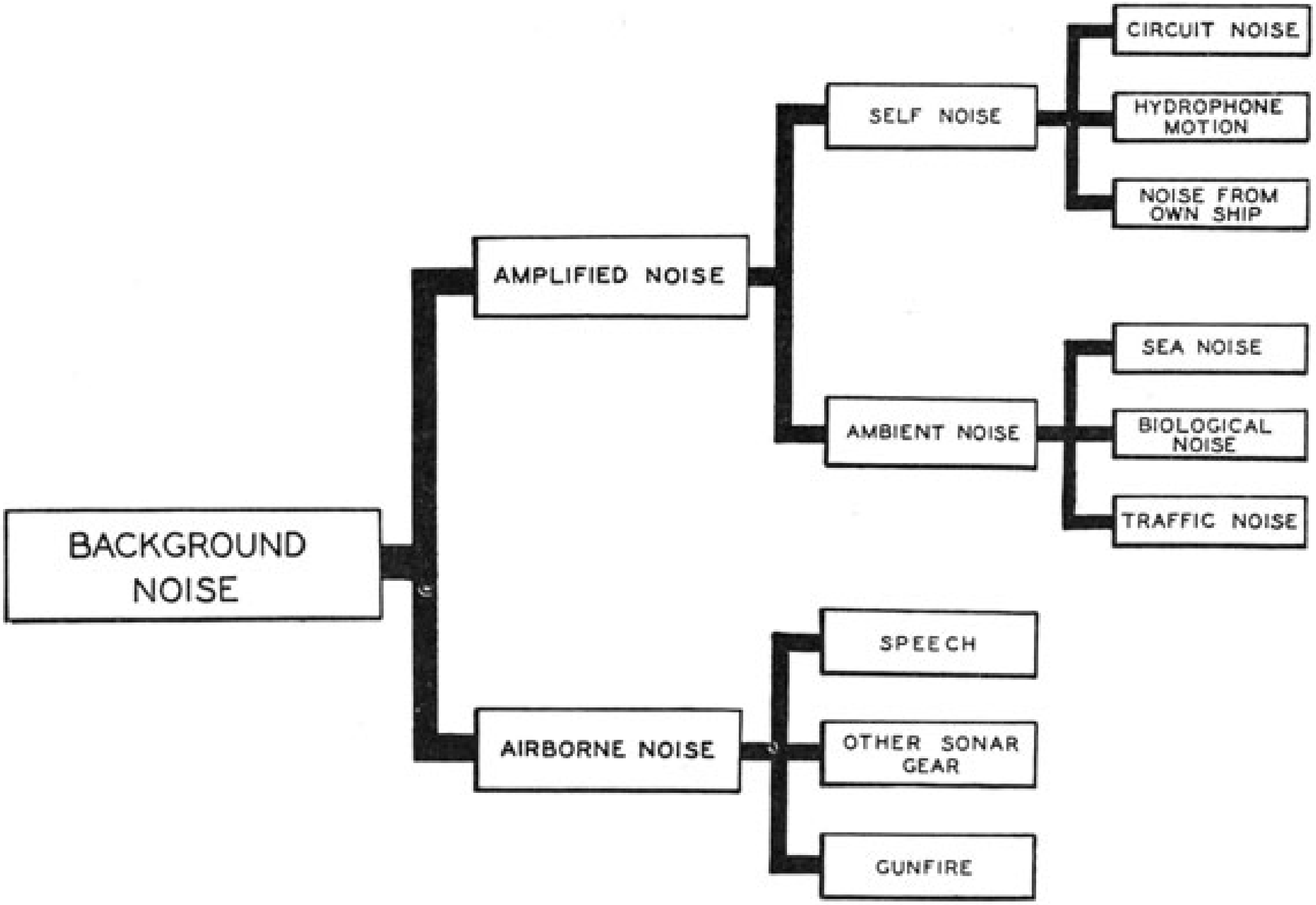


Figure 3-3. -Classification of background noise.

TABLE 5. -*Over-all Levels of Amplified Noise (0.1 to 10 kc)*

Types of noises	Decibels
Self-noise:	
Circuit noise	-30 to 0
Submarine self-noise	0 to 20
Surface vessel self-noise (DD or DE) 10 to 25 knots	5 to 40
Ambient noise:	
Sea noise:	
Deep sea	-5 to 6
Near surface	-17 to 9

Table 5 is a summary of the average values of background noise of all kinds. This table, which gives some interesting information regarding the intensity of noise made by fish, will be referred to from time to time.

SOUNDS PRODUCED BY OBJECTS IN THE SEA

Biological Noise

Surprisingly large numbers of species of marine life produce sounds of various sorts. They are mostly crustaceans and vertebrates. Biological noise is an important factor in limiting listening ranges in

Biological noise:	
Snapping shrimp	5 to 7.5
Croakers	36 (max.)
Porpoises	40 (max.)
Evening noise	8.5 (max.)
Traffic noise (includes sea noise)	0 to 22

wave motion at the surface of the water; (2) biological noise, caused by many species of marine life; and (3) traffic noise, which exists when many ships operate at the same time, such as in a harbor. The noise of fish and marine life is not always undesirable but in the detection of ships or submarines is usually a source of trouble. Because this type of noise is rather peculiar, it will be discussed in some detail.

shallow water only in tropical and subtropical regions. To discuss the complicated subjects conveniently, it is customary to group the various sounds from marine life into three categories, which in the order of their importance from an operator's viewpoint are (1) shrimp noise, (2) periodic fish choruses or croaker noise, and (3) miscellaneous biological noise.

Early in World War II it was observed that as a listener approached shallow water, the ordinary ambient noise was sometimes replaced by sounds resembling the sizzle of frying fat. As he came closer to the shore, he noticed that the sound approximated the crackle of burning twigs or the

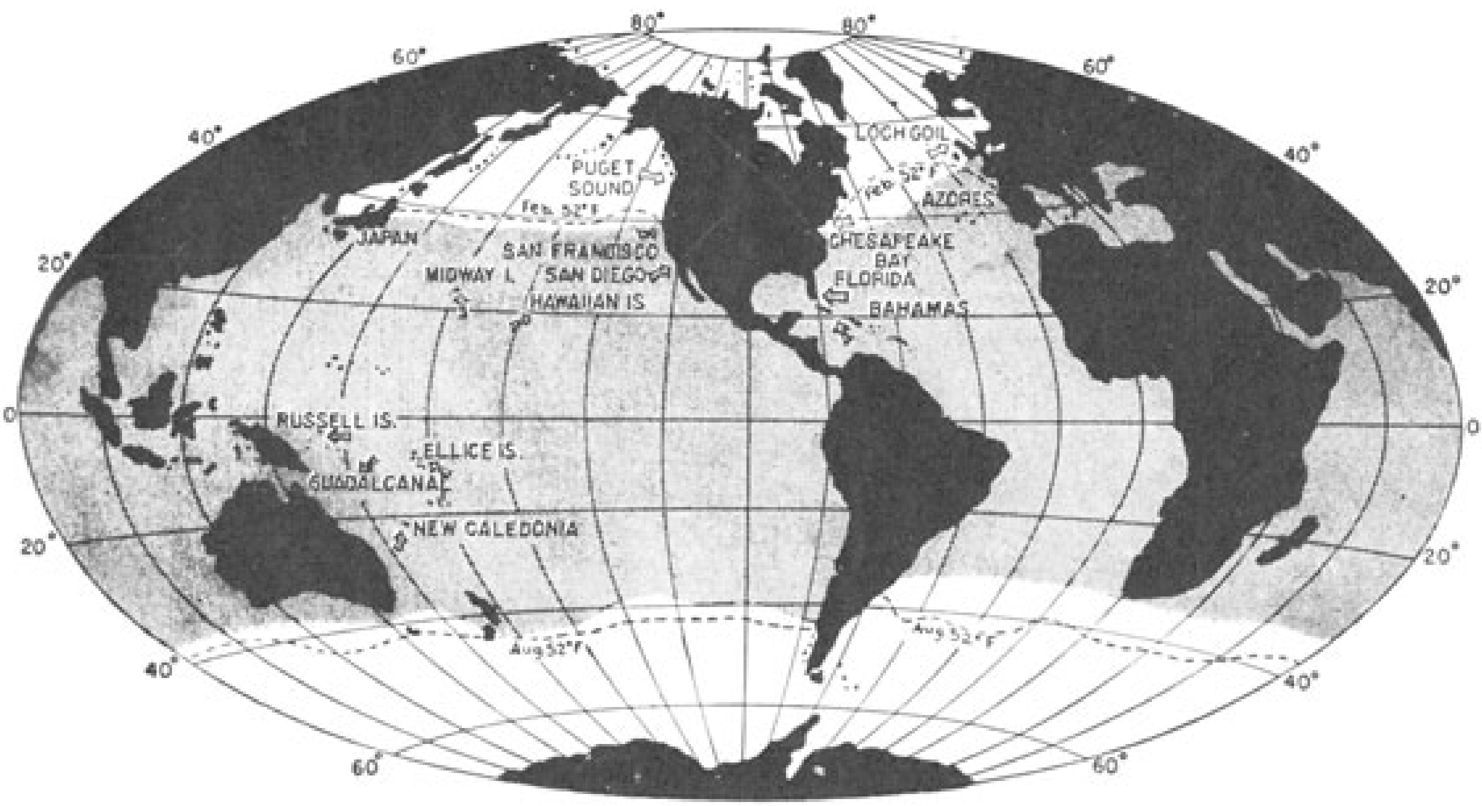


Figure 3-4. -Distribution o snapping shrimp.

239276°-5-5

crashes of static noise heard in a radio receiver. This noise was encountered only in tropical and subtropical regions, and was more common over boulder-strewn or cobble-strewn bottoms. It was sometimes confused with noise due to surf. Investigation discovered the source of this noise to be colonies of certain species of snapping shrimp (not to be confused with the ordinary edible species) that close their pincers with a loud audible click, similar to that caused by snapping a fingernail. The rate at which a single shrimp produces clicks and the reason for this activity are not known. The combined activity of hundreds of thousands of shrimp is required to produce the observed sizzle.

The chief habitats of these shrimp are in coral formations and on rocky sea bottoms where the water is less than 30 fathoms deep. Few are found on mud or sand bottoms. The map in figure 3-4 shows that they are widespread throughout tropical and subtropical regions of the world. In this figure, shaded areas show regions where shrimp occur when water depth and bottom are favorable.

Shrimp noise is a serious masking noise in listening, both because of its intensity and because of its spectral distribution. Although it has a

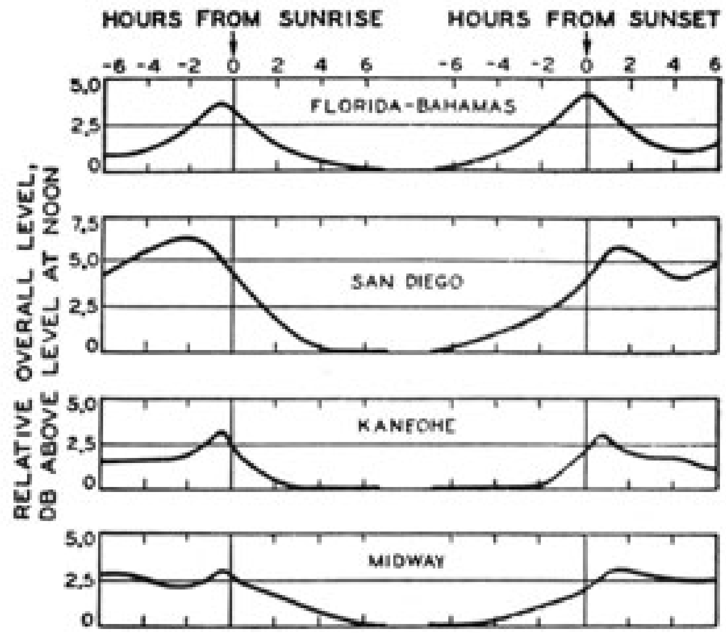


Figure 3-6 -Diurnal variation of shrimp noise, over-all level at various locations.

measured frequency range of from 1.5 to 45 kilocycles the main components lie between 1.5 and 20 kilocycles. The spectrum level at 10 kilocycles may be of the order of -39 to -29 db, as can be seen from figure 3-5. In this figure the dots indicate average values; the dotted curves show the spread of the spectrum levels. It is evident that shrimp noise is a serious complication in both sonic and supersonic listening.

Shrimp noise is remarkably constant throughout the year. There is a small diurnal variation-the noise is from 2 to 6 db higher at night than in daytime, small maxima occurring about 1 hour before sunrise and about 1 hour after sunset. (See figure 3-6.)

The chief noise makers among fish are certain species of croakers and drumfish, which are common, especially on the Atlantic coast. An individual croaker emits sounds resembling 4 to 7 rapid blows on a hollow log.

At certain periods of the year large schools of croakers infest certain localities. In the Chesapeake Bay the croaker season extends from May to July. During this season there is an evening chorus of

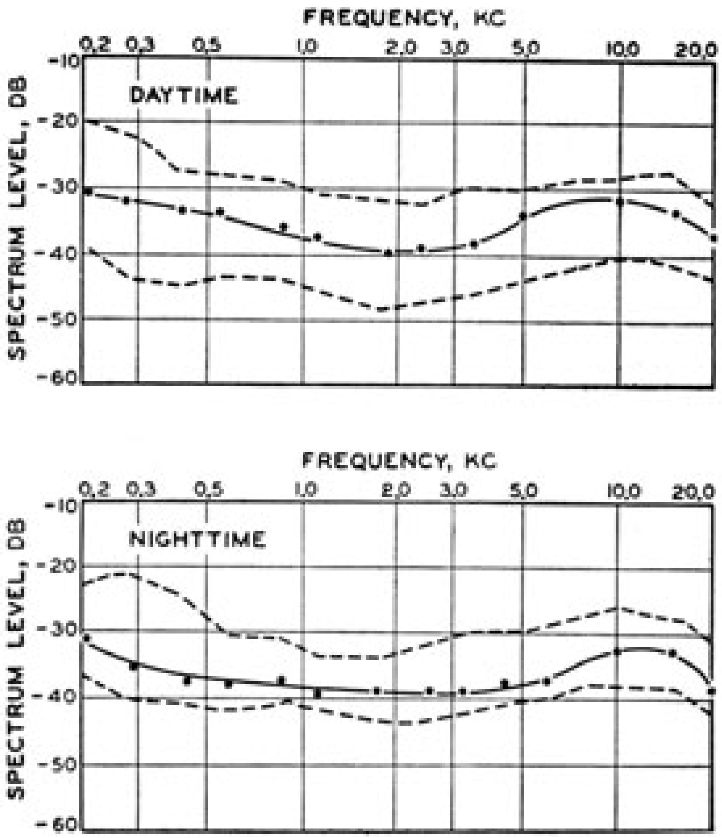


Figure 3-5 -Spectra of shrimp noise for daytime and nighttime.

croaker noise lasting several hours, with a peak just after sundown. Over-all levels of croaker noise showing seasonal and diurnal variation are shown in figure 3-7.

The spectrum levels of a sample of croaker noise are shown in figure 3-8. The solid curves show the difference in average level between early evening and the period after midnight during July.

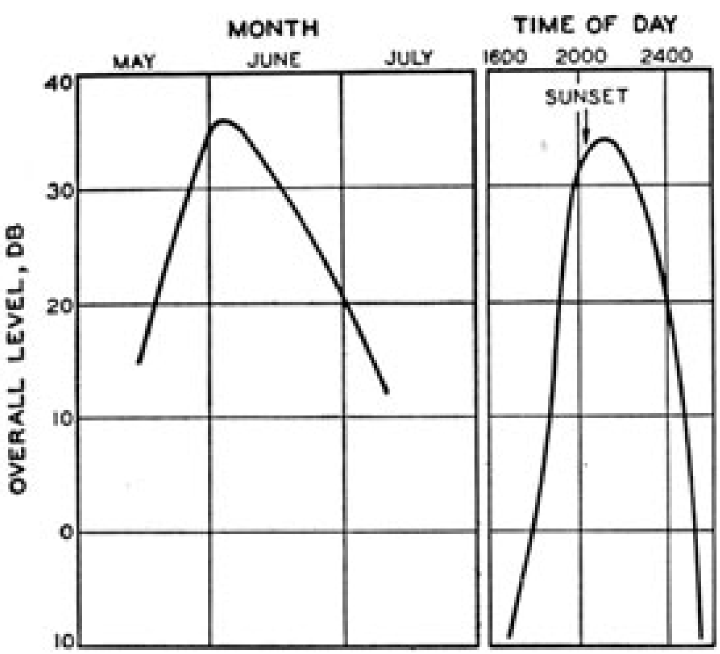


Figure 3-7 -Seasonal and diurnal variation of over-all levels of croaker noise.

The dotted curve is the average spectrum for early June. When it comes, croaker noise may

off the harbor entrance have been devised to ensure protection of harbors against sneak attacks by enemy submarines.

Traffic noise is essentially variable, but a certain periodicity can be expected. Measurements made in New York Harbor and its approaches are shown in figure 3-9. Curve *A* shows the spectrum level of the noise in the harbor in the daytime, and curve *B*, the average levels measured in upper Long Island Sound near the ship lanes. Curve *B* is about 9 db below the harbor level at all frequencies. For comparison, the curve of sea noise for sea state 2 is included as curve *C*. In the region of sonic frequencies the harbor noise is from 10 to 18 db above this level. Over-all sound levels (0.1 to 10 kc) for the noise in the harbor itself is about 16 db, compared with 6 db in the harbor approaches and 0 db for water noise with sea state 2.

completely mask desired signals, for the frequency range of croaker noise lies almost entirely below 1 kc, the region where the most prominent components of ship sound occur.

In and near busy harbors the ordinary sea noise and biological noises are overlaid with the sounds associated with the movements of ships, especially small high-speed craft, and by the noise of industrial operations on the beach. Listening in harbors thus becomes extremely difficult; hence installations

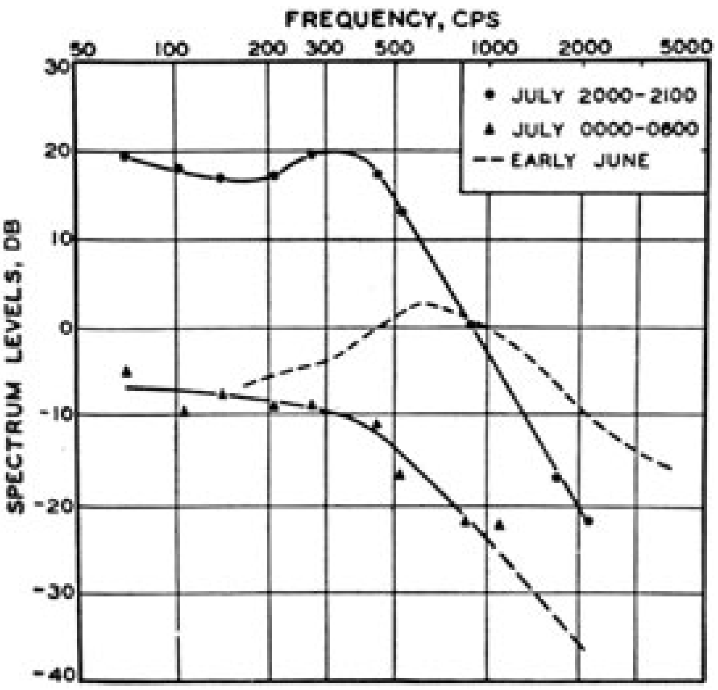


Figure 3-8 -Spectra of croaker noise.

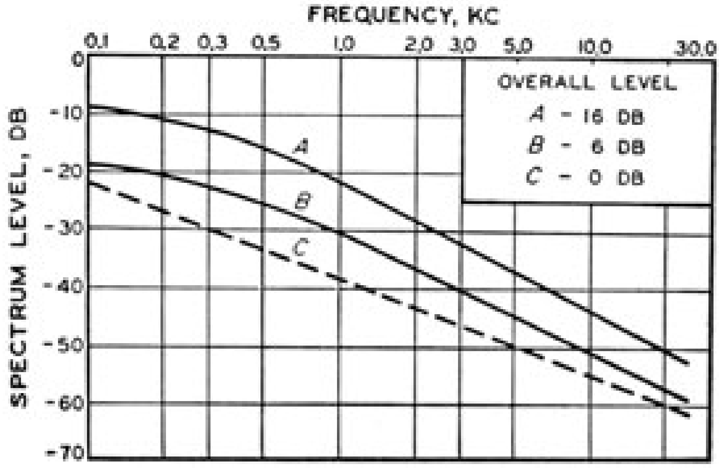


Figure 3-9 -Spectra of traffic noise in New York Harbor and its approaches during the daytime.

Nighttime levels of ambient noise in the approaches to New York Harbor are shown in figure 3-10, with a curve showing average daytime levels added for comparison.

Submarines

From the standpoint of antisubmarine operations, a knowledge of the sound output of submarines is needed for the prediction of maximum listening ranges. The design of listening gear, in particular the choice between sonic and ultrasonic devices, depends on the spectrum of the sound to be detected.

From the standpoint of submarine operations, it is important to know the relative sound output of various submarine maneuvers, so that evasive action is not nullified by excessive detectable

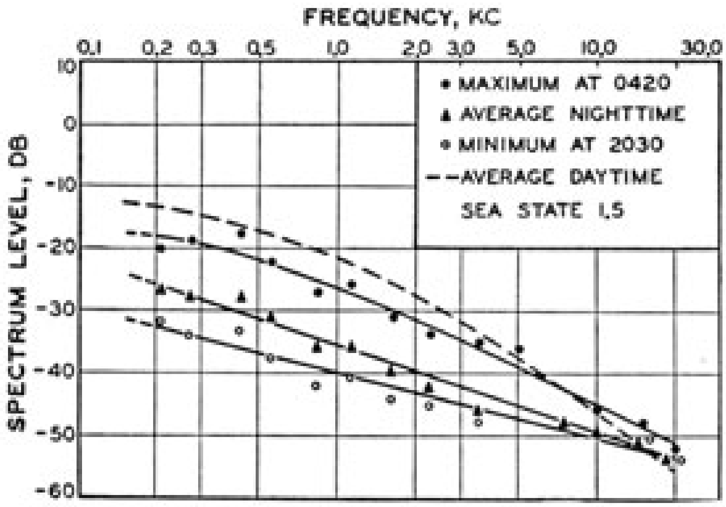


Figure 3-10. -Same as figure 3-9 but for nighttime.

sound. The problem of noise control, and the design of propellers, engines, and auxiliaries, all demand measurements of sound output.

The machinery of the submarine is extremely diversified and complicated. The submarine has more than 50 auxiliaries, all of which are potential sound sources. Figure 3-11 lists a few of these sources, shows the source levels that have been proposed as best naval practice, and gives the maximum permissible limits.

In general, these sounds have a continuous spectrum, with a maximum at low frequencies. Sometimes, however, the machinery produces a strong line spectrum that is superimposed on the continuous spectrum.

Propeller sounds are of two general kinds- (1) *singing*, due to vibrations of the propeller blades, and (2) *cavitation*. Cavitation sounds are the most important of all submarine sounds. Vibrations of the propeller blades may be due to faulty design or manufacture and are generally not difficult to eliminate.

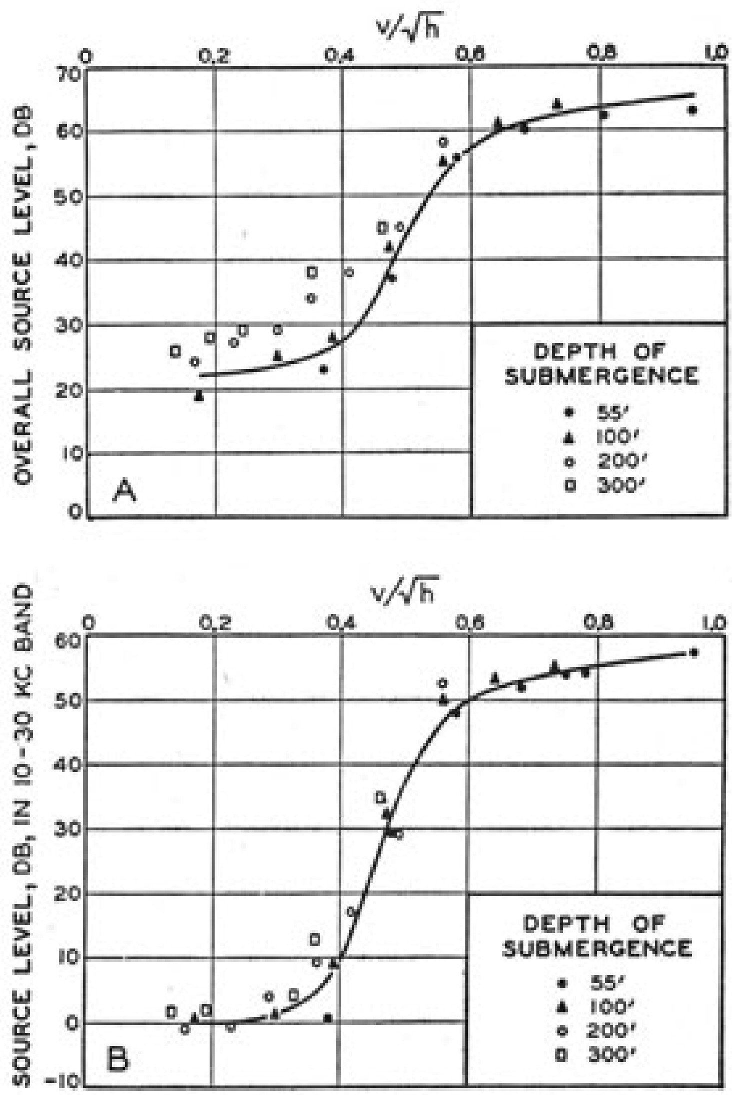


Figure 3-12 -Dependence of over-all source levels of submarine sounds on depths of submergence *h* (feet) and speed *V* (knots). A, 0.1-kc to 10-kc; B, 10-kc to 30-kc.

Cavitation results when the propellers turn so rapidly that the water does not close in behind the

blades. Thus, a stream of bubbles resembling those in a boiling kettle is formed. These bubbles may be caused by reduced pressures on the backs of the propeller blades or by vortices at the tips of the propeller blades.

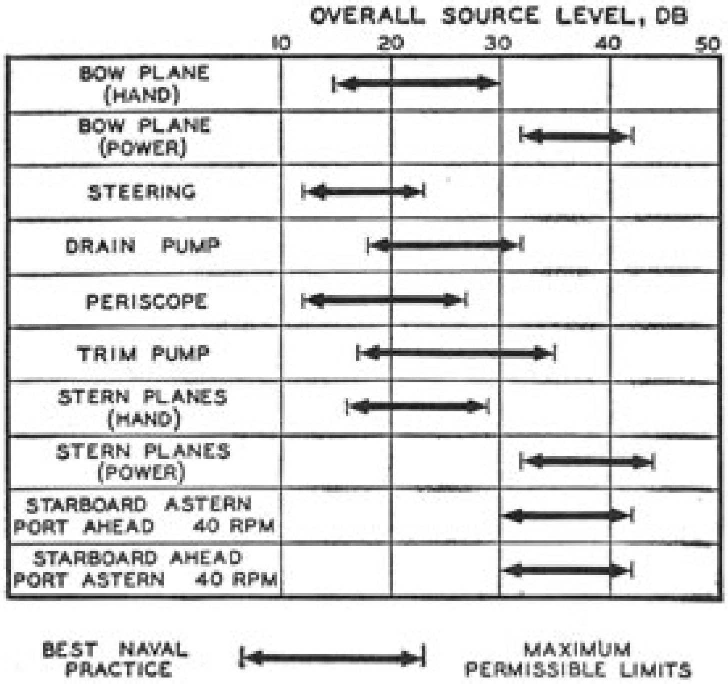


Figure 3-11 -Suggested limits of over-all sound level of several auxiliaries on submarines, and the levels representing best naval practice.

The steep rise between the value of 0.4 and the value of 0.6 for $V/h^{1/2}$ is due to cavitation (figure 3-12). The smooth curve is drawn on the assumption that the speed at which cavitation occurs is inversely proportional to the square root of the hydrostatic pressure. Figure 3-12, A, plots the levels measured in the 0.1-kc to 10-kc bandwidth; figure 3-12, B, the levels in the 10-kc to 30-kc band. Acoustically, tip cavitation appears to be much more important than blade cavitation. This condition may exist because blade cavitation has a more serious effect on propeller thrust and is usually prevented by the designer of the ship.

Besides these two main sources of submarine sounds, there are some minor sources, such as splashing of water at the bow and in the wake when the submarine is at the surface; when submerged, the fittings of the vessel, such as handrails, may be set into vibration by the turbulent flow of water past them. These sounds are considered to be of small significance

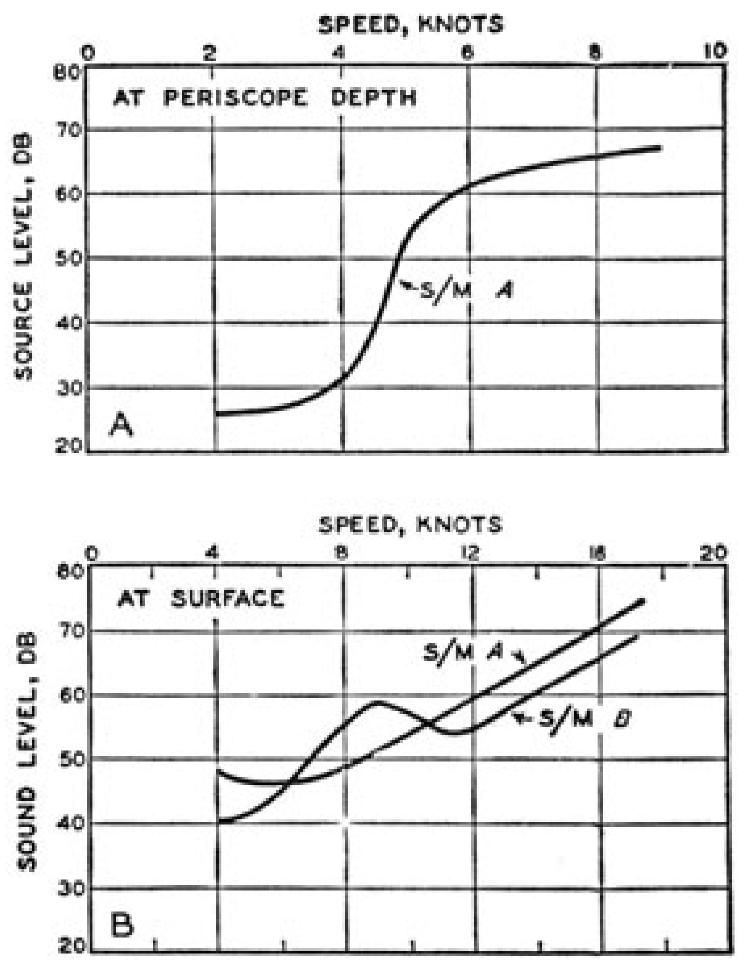


Figure 3-13 -Over-all source levels of submarine

compared with those due to cavitation.

The activities of the crew are a source of incidental sound. It is interesting that, according to some British measurements, over-all source levels of from 45 to 50 db may be produced by dropping a wrench or by the use of the engine-room telegraph-levels comparable to those produced by the submarine itself under conditions of evasive operations. The transitory character of such sounds makes them comparatively unimportant, except when the submarine is evading detection by an alert enemy.

The sound output of a submarine varies widely with the size and type of submarine. For a given submarine it varies with speed and operating conditions. If the submarine is submerged, its sound output at a given speed decreases as the depth increases.

The over-all source level may range from about 40 db under evasive conditions to more than 75 db at top speeds. An average based on a large number of measurements gives the following values: (1) Running submerged at 6 knots, or on the surface at 12 knots, the over-all source level is about 72 db; (2) at top surface speeds, the over-all source level is about 77 db.

The dependence of the over-all source level on speed is shown for two submarines in figure 3-13.

sounds. A, Submerged variation with speed; B, two submarines, surface operation, illustrating the variability between ships.

In figure 3-13, A, the over-all source level is plotted against the ship speed for a submerged submarine, and in figure 3-13, B, for two submarines operating at the surface.

The variability of source level from ship to ship is indicated by the curve of submarine B in figure 3-13, B. The values of source levels of various submarines may vary by as much as 15 db under identical operating conditions.

The curve pertaining to operation at periscope depth is typical of ship sounds in general. At very low speeds the source level is quite low. At a certain critical speed-in this case 4 knots-the sound output increases very rapidly with speed, so that an increase of 2 knots is accompanied by an increase in the source level of 30 db. If the speed is increased beyond 6 knots, the curve levels off.

This abrupt increase in the sound output at the critical speed is due to cavitation, which is related to many factors but chiefly to the shaft rate or speed and to the hydrostatic pressure. If other

factors remain constant, the speed at which cavitation occurs is inversely proportional to the square root of the static pressure. Hence the sound output at a given speed is less when the submarine submerges to greater depths. This fact is shown by figure 3-12, in which over-all source levels are plotted against $V/h^{1/2}$ where V is the speed in knots and h is the total hydrostatic pressure head. The value of h is calculated from $h=33+d$, where d is the depth in feet and 33 feet is the head of sea water equivalent to 1 atmosphere. The experimental points fit the theoretical curves fairly well.

The speed required for cavitation to set in is, in general, higher for submarines of new design because of a persistent effort to decrease the sound output of American submarines. It has been decreased, on the average by about 20 db; however, a few submarines still produce prominent and undesirable single-frequency tones below 1,000 cycles per second. There is considerable evidence that these sounds originate almost entirely in the reduction gears.

The relation between sound level and speed of a submarine is quite different for surface operation. Figure 3-13, B, shows that the increase in source level of submarine A is gradual, and does not show the abrupt rise due to cavitation that is observed with submerged operation. The higher levels associated with surface operation are attributed to the Diesel engines used for operating on the surface; the electric drive is considerably more quiet. The hump shown in the curve for submarine *B*, figure 3-13, B, is caused by a singing propeller.

Figure 3-14 gives the spectrum of a submarine running at 6 knots at periscope depth or at 12

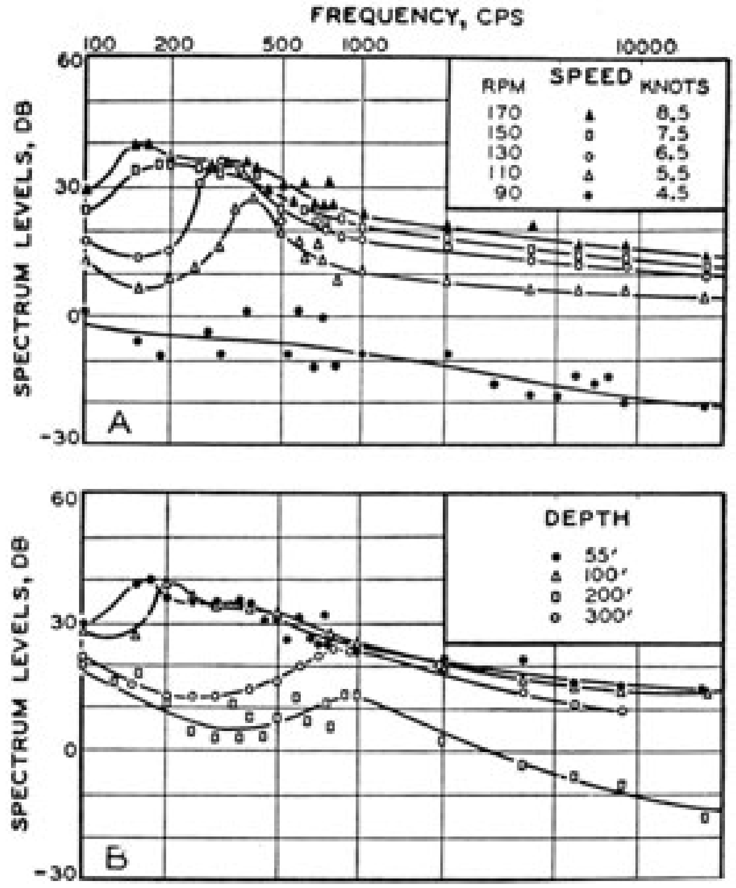


Figure 3-15 -Spectra of individual submarines. A, The variation of spectra with speed of submerged submarine; B, effect of increasing depth on the spectra.

knots on the surface. These values are the average of a large number of measurements. It must be borne in mind that there is a great spread in individual measurements, and thus the sounds from a given submarine may deviate decidedly from the values in the figure.

Figure 3-14 shows that the intensity of submarine sounds decreases rapidly with the frequency; the drop in level is about 6 db per octave on the average. In other words, the spectrum level is about 20 db higher at 100 cycles per second than at 1,000 cycles per second and this same proportionate variation continues at least until 30 kilocycles. As a result, the over-all level is largely determined by the lower frequencies.

If the threshold of listening gear were independent

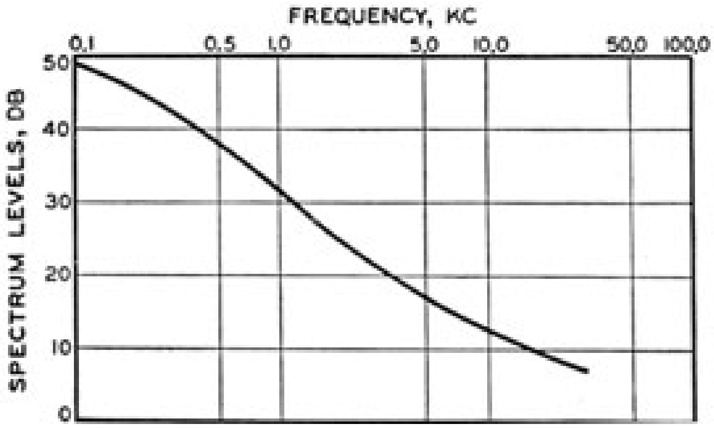


Figure 3-14. -Average spectrum of a submarine running at 6 knots at periscope depth or at 12 knots on the surface.

of frequency, sounds with such a spectrum would be much more readily detected with sonic than with ultrasonic devices. However, the threshold also decreases with increasing frequency, especially for gear mounted on a moving surface vessel. Until recently this factor has tended to nullify the advantage of sonic listening. On

---

sailing vessels, sonic listening retains its advantage, especially if the auxiliaries can be periodically shut down for listening. An effective antisubmarine watch can thus be maintained from such vessels. The same is true of bottom-mounted hydrophones and sonobuoys, both of which use the sonic band.

Sound-level spectra of individual submarines are shown in figure 3-15 and figure 3-16 for various operating conditions. Figure 3-15, A, shows the effect of increasing speed on the sound-level spectrum. A characteristic feature of these curves is a peak at low frequencies, and a tendency for this peak to occur at lower frequencies as the speed increases. This behavior is ascribed to cavitation effects. It is thought that higher propeller speeds produce progressively larger bubbles. The resonant frequency of a bubble is inversely related to its diameter, and thus an increase in speed results in the production of sound of a lower frequency.

The exact position of these peaks also varies from submarine to submarine. Consequently they do not show on the average curve of figure 3-14. Even the peaks of these submarines lie well below the average curve for frequencies of less than 1 be considered as the effective source of the radiated sound. There is reason to believe that at periscope depth the engine room is the principal source of sounds at very low speeds, whereas at speeds above 3 knots the propeller is chiefly responsible. however, even at high speeds the engine room may contribute materially to the sound at frequencies below 150 cycles per second. During surface operations the propeller and wake are probably the principal sources of sound at practically all speeds with electric drive. With Diesel drive the engine room is the main source at low speeds and a material contributor at all speeds.

The sounds from submarines are radiated in such a way as to produce approximately a uniform sound field at a distance of several ship lengths from the source. Some observers report a slight decrease in the sound level in the region within 10° or 20° on either bow; at 200 yards this decrease is from 2 to 4 db. A similar shadow astern of the ship has been reported. This shadow is ascribed to the wake.

Surface Ships

The sounds emitted by surface vessels may provide considerable information to an experienced sound operator aboard a submarine. Various forms of

kc.

Figure 3-15, B, shows the effect of increasing depth on the sound-level spectrum. The peaks tend to shift toward higher frequencies with increasing depths. The increase in hydrostatic pressure with depth probably reduces the size of the cavities formed at a given speed and thus results in a higher resonant frequency.

Very little is known concerning the location of the particular point, or points, on the ship that can

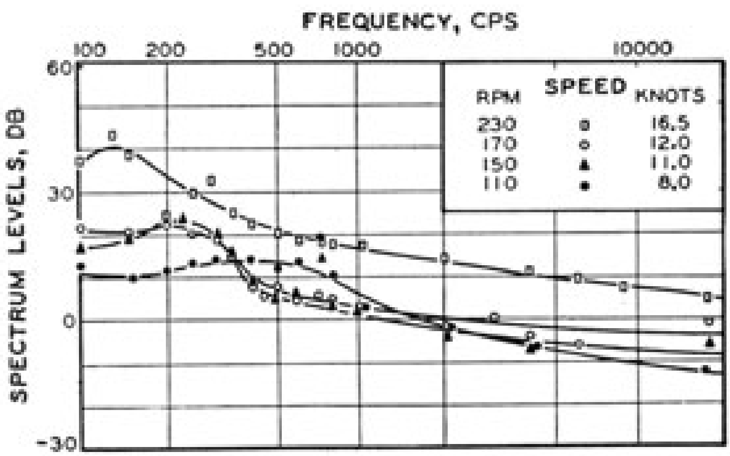


Figure 3-16 -Variation of spectra of individual submarines with speed in surface operations.

underwater mines are detonated by a ship's sound. Ship sounds vary greatly in intensity and spectrum from ship to ship and from one class of ship to another. For a given ship sound intensity and spectrum vary with speed.

From the viewpoint of defense, every ship that is likely to enter water harboring hostile submarines obviously would benefit by an analysis of its own sound output. Such an analysis would disclose the existence of any revealing single-frequency components. These undesirable components are due to causes that can be remedied easily. The analysis also would make possible more accurate estimates of the range at which a ship is apt to be detected by an enemy submarine.

The extreme values of observed over-all source levels range from about 50 db for launches and small auxiliary craft at low speeds to 110 db for battleships at 20 knots. The 110-db value is approximately the source level of a standard sonar projector. The average over-all source levels of submarines range from about 30 to about 75 db.

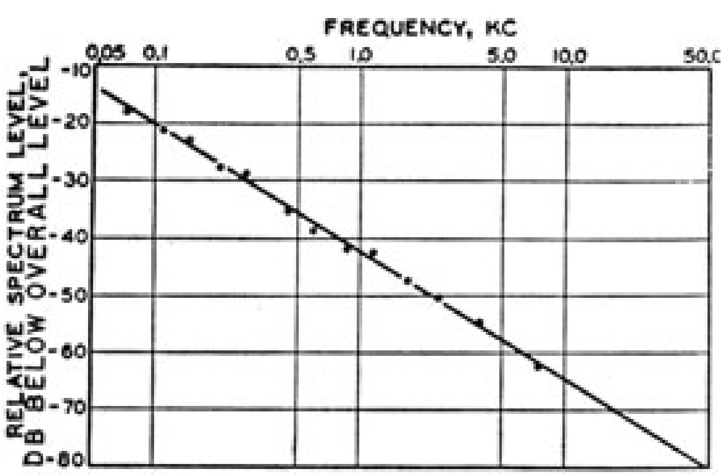


Figure 3-17 -Spectra of surface ships.

Besides being affected by the speed of the vessel, the over-all source level is a function also of the load or displacement of the ship.

submarines, the chief sources are the screws, where cavitation produces the sound, and the hull, which transmits the vibrations of the machinery and engines.

Single-frequency components due to propeller singing or to vibrations of the propulsion machinery are common. Ordinarily such sounds occur below 1 kilocycle, but sometimes these single-frequency components are encountered well above this frequency.

Figure 3-17 shows the average spectrum-frequency distribution of sounds from a large number of surface ships. The data on which this figure is based

The sources of ship sounds are extremely diversified, and a given source may change its sound output with ship speed. Hence ship sounds are variable and complex and are distributed through the whole range of frequencies. As with were the average measurements made on 52 ships comprising 12 different types of warships and commercial vessels. The ordinates on the graph are the values of relative spectrum levels-that is, of the spectrum level minus the over-all level (0.1-kc to 10-kc). These differences are averaged for all types of ships in order to obtain the graphs. Because the total spread of the measurements on the individual ships was considerable, due allowance for this spread must

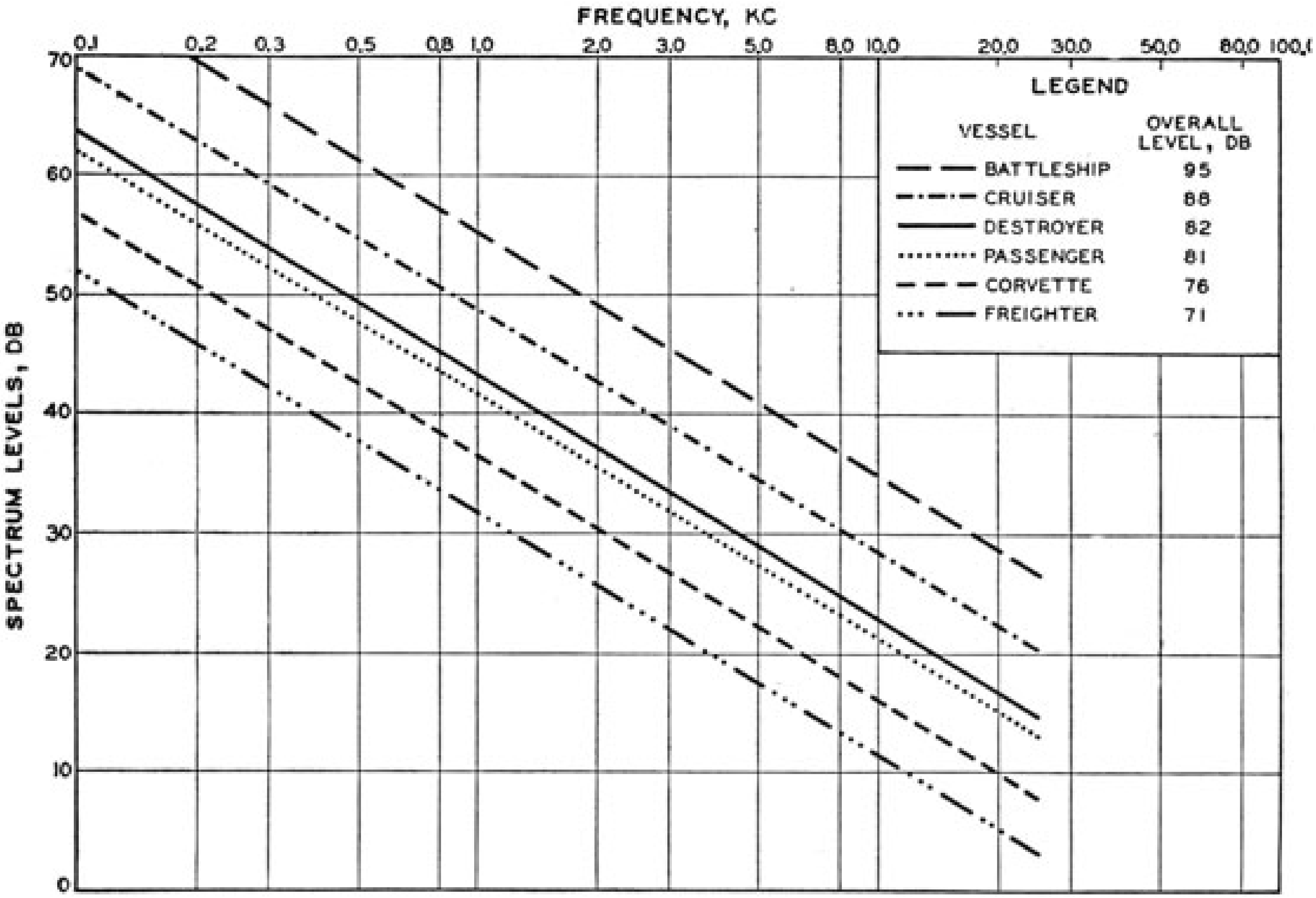


Figure 3-18. -Average spectrum levels for six different classes of ships.

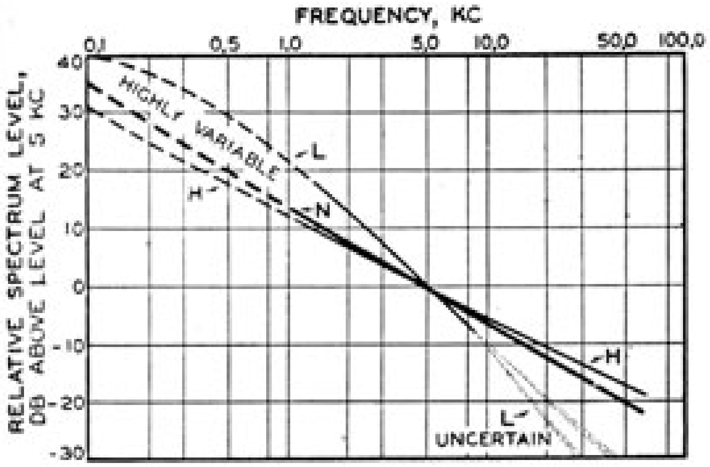


Figure 3-19 -Effect of varying speed on spectral distribution.

be made when using data from this graph and the following graphs.

The level of the sound decreases with increasing frequency at a rate of 7 db per octave. This relation is similar to that shown in figure 3-14 for submarines. Spectra of the different ships vary in average slope from about 5.5 to about 8.6 db per octave. Figure 3-18 shows average spectrum levels for six different classes of ships at normal cruising speeds. The average over-all levels also are indicated.

Figure 3-19 illustrates the effect of varying speed on a ship's spectral distribution. Curve *L* represents the average spectrum at low speeds, curve *H* that at high speeds, and curve *N* that at normal cruising speeds. At very low speeds the chief source of sound is the machinery, and all the machinery contributes materially. Much of the sound from this source is concentrated at the lower frequencies; therefore in this region the spectrum is highly variable, as was previously noted with submarines.

The variability of the spectra in the lower frequency region may be ascribed again to cavitation, which is the chief source of ship sounds at all but the lowest speeds. The sound due to cavitation has a continuous spectrum, whereas

At high speeds cavitation may introduce components in the ultrasonic region, as shown by curve *H* in figure 3-19.

The sound emitted by ships has very little directivity, particularly in the sonic region of frequencies. Average directivity patterns for 15 freighters for the low frequencies (200 to 400 cycles per second) are illustrated in figure 3-20, where sound levels are exhibited as contours-lines joining points of equal intensity. The levels were measured with a bottom-mounted hydrophone.

Contours are somewhat difficult to reconcile with the fact that many ships have two dominant sources of sound, one at the engine room and the other at the screws. In large destroyers these two sources are of equal level at about 12 knots. At 8 knots the engine room is the dominant source, whereas at 16 knots the screws are the dominant

machinery sound generally is more likely to consist of many discrete components closely spaced. Above approximately 1 or 2 kc the spectral slope of cavitation sound is very nearly -6 db per octave; but in the region of lower frequencies there is usually a peak (figure 3-15). The frequency at which this peak occurs depends on various factors related to the type and .size of ship and its speed, and thus may provide some information tending toward identification of the vessel.

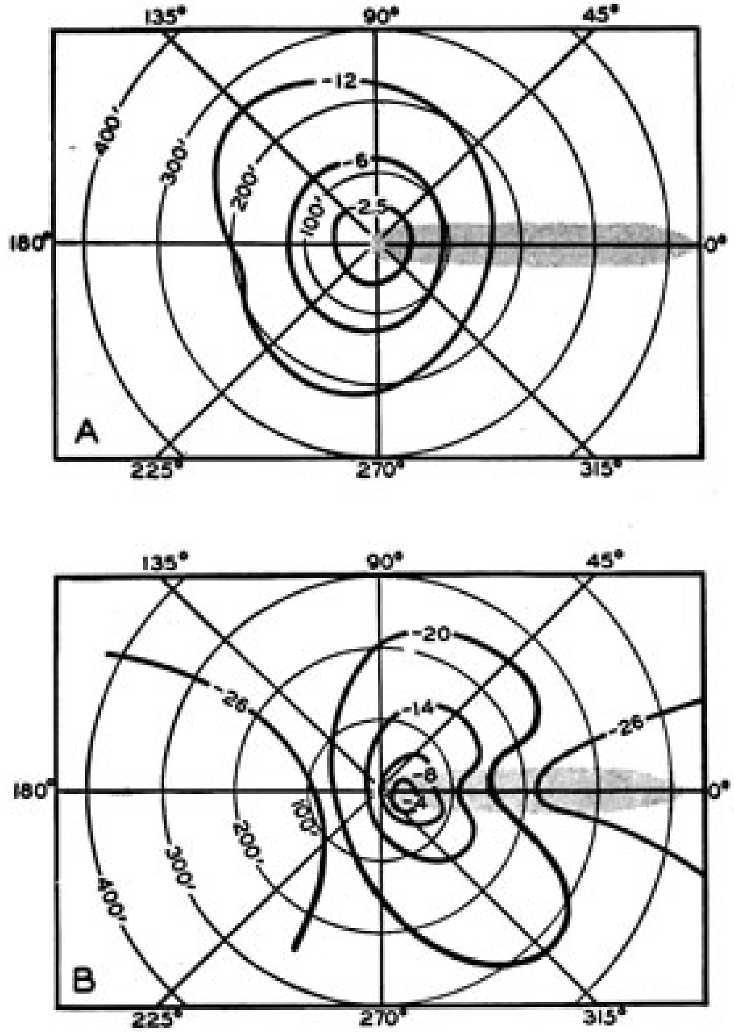


Figure 3-20 -Contours showing the average directivity of ship sounds. A, Average patterns for 15 freighters for low frequencies (200-400 cycles per second); B, contours of sound levels for a typical freighter at 8 knots. The outline of the ship is indicated by the shaded area.

source. In Liberty ships, however, the two sources are of about equal level at all speeds. The dominance of the propellers as the source of sound for the 15 ships shown in figure 3-20, A, possibly indicates that Liberty ships are not typical of all freighters.

If the source of sound from a ship is concentrated at the screws or over a small part of its hull, the audible sound is independent of direction except for the shadow effect of the hull and wake. This effect is illustrated graphically in figure 3-20, B, which shows the contours of pressure levels for a typical freighter cruising at 8 knots. The outline of the ship is shown by the shaded area. The shadow and screening effects are highly variable from ship to ship. These variations and the variable distribution of the sound sources make it difficult to generalize about the sound distribution. It is probable that for large ships the sound-pressure level 400 to 500 feet ahead or astern of the main source of sound is 5 to 10 db below the level at the same distance abeam.

Time Patterns and Propeller Beats

RHYTHMS AND OTHER TIME PATTERNS

The necessary prerequisite for the detection of a ship or submarine is that its sound have sufficient intensity at the hydrophone to be heard above the background noise. Because the level of background noise usually varies in an irregular manner, a rhythmic sound having a periodic pattern of beats, may be more readily recognized than a nonrhythmic one.

Moreover, intensity alone conveys no information other than that something in the neighborhood is making a noise. Additional information about the source is obtained from the spectrum (high or low pitch) and from any rhythm that is inherent in the sound.

The propeller sounds of a large ship, although produced by cavitation, usually pulsate periodically. In some ships, the beat may be unaccented and occur once per propeller revolution (shaft frequency). Other propeller sounds pulsate several times per revolution; a three-blade propeller gives 3, and a four-blade, 4, beats per revolution (blade frequency). If the beat is unaccented, it is difficult to determine which frequency is involved. However, one blade is they are relatively large; they are often called *fading*. Rhythms are most easily heard and counted when the beats occur two or three times a second. At high rates, counting becomes difficult; with practice, it can be done by counting every third or fourth beat.

When the frequency becomes greater than about 15 or 20 cycles per second, the individual beats are no longer heard. The rhythm is then heard as a "flutter" or "tremolo". Frequencies much above 100 cycles per second are not recognized as periodic, but as a pitch that is inherent in the sound.

SINGLE-FREQUENCY COMPONENTS

Audibility

Previous discussions in this chapter have pointed out that ship sounds in general have continuous spectra-that is, (1) the emitted sound energy is distributed over a wide range of frequencies, and (2) on the average, the distribution of the energy over the frequency range follows a fairly simple pattern-a decrease in the sound level of about 6 db per octave increase in frequency.

Mention has been made, however, of the occurrence in ship sounds of relatively pure tones of audible

often noisier than the others, resulting in an accent repeated at shaft frequency. In favorable cases, therefore, both the number of blades and the propeller rpm can be determined. These items partially identify the class of ships, and certainly differentiate its sound from various intermittent background noises.

Perception of Time Patterns

The manner in which fluctuations in sound level are heard depends on their rate or frequency. Very slow changes in level are not perceived unless

frequency. On a spectrum plot an absolutely pure tone would be one-dimensional having sound level but no frequency width. A spectrum composed predominantly of such discrete components would be a line spectrum. Actually the so-called single-frequency components comprise a relatively narrow band of frequencies; but if the width of this band is smaller than the width of the band that can be resolved by the ear, the single-frequency components will have a definite

---

pitch. It is in this sense that the terms "single-frequency component" and "pure tone" are used.

The ear very readily detects pure tones against a background of complex noise. This detection is possible because the ear is a very efficient analyzer of comparatively high selectivity, and because a pure tone has a distinctive quality that contrasts strongly with random noise, which has no definite pitch. These characteristics make it possible for the ear to detect a pure tone in the audible region even when its sound level is considerably lower (sometimes as much as 20 db) than the over-all level of the background noise.

Tests have shown a pure tone can be heard when its level is at least equal to the level of the background noise in a band of a certain width at the frequency of the tone. The width of the band depends on the frequency. These critical bands are from 30 to 50 cycles per second wide for tones of between 100 and 1,000 cycles per second. This fact is an indication of the great effectiveness of the ear in discriminating against random noise.

Time Patterns

Pronounced rhythmic time patterns sometimes occur in single-frequency components originating in propeller vibrations. Also, many single-frequency components have their source in reduction gears.

The extreme audibility of single-frequency components, as compared to sounds of continuous spectrum, introduces complications in the techniques of sound measurement. For example, suppose the over-all level of a moored submarine with its motors secured is measured. It has a continuous spectrum of certain over-all level. The motor may produce a pure tone that increases the audibility of the submarine's sound very materially, but may scarcely affect the over-all level.

Frequency Considerations in Listening

In the over-all problem of detection by listening, two general classes of systems can be distinguished. One class includes those systems with a listening band that falls in the ultrasonic region and an output that is made audible by a heterodyne change of frequency. The other class has its listening band in the audio frequencies and does not need a heterodyne stage to make the output perceptible.

SONIC LISTENING

Sonic listening depends on the sources of sonic sounds. These sources are surface vessels, submarines, torpedoes, explosions of depth charges, and the echo-ranging signals of other vessels. Cavitation sounds have a comparatively continuous spectrum, the level of which falls off about 6 db per octave on the average. They are sufficiently uniform to make it possible to determine the cavitation spectrum of a given class of ship at a given speed by taking a single measurement at some frequency-say 1 kc or 5 kc. Enough measurements on cavitation sounds from various sources have been made to enable the prediction of their level for any class of ship at any speed within about 5 db.

It is not so easy to predict the level of machinery sounds, which are the dominant source of low-frequency sound (less than 1 kc) at low speeds. These sounds have very complex and irregular line spectra and differ widely among different ships. They are heard as squeaks, rumbles, groans, and whines.

The spectra of the various types of ambient noise that are encountered in listening have been discussed. Ambient noise is the limiting factor when the listening hydrophone is stationary, provided the sea state is greater than 1 or 2. For a sea state of less than 2, the over-all level of ambient noise drops below 0 db and thus approaches the over-all level of circuit noise, which ranges from -30 to 0 db. In this case, the circuit noise may be limiting. Shrimp noise is usually negligible at lower sonic frequencies.

The data for transmission loss in the frequency range of from 200 to 2,000 cycles per second can be schematically summarized (figure 3-21).

At ranges less than a few hundred yards, the transmission loss, H, is variable because of the interference between direct and surface-reflected sound. This condition is indicated by the double hatching in the figure. Beyond this variable region, the transmission loss increases rapidly out to about 2,000 yards. The frequency is a determining factor in this region. The low frequencies suffer a greater loss than the high frequencies.

Downward refraction in the upper layers causes this loss to occur at shorter ranges. The single hatching on figure 3-21 shows the region of the rapidly increasing loss.

Beyond this region bottom-reflected sound is dominant, and the transmission loss remains constant out to about 20,000 yards. The magnitude of this loss and the range at which it begins depend on the depth of water. A value of 80 to 85 db appears to be relatively independent of thermal conditions but increases slightly with the hydrophone depth. This value is also subject to irregular fluctuations of considerable magnitude, but they do not appear to bear any systematic relation to the range.

At very long ranges the transmission loss must again increase, but there is very little data to indicate the rate of increase.

The fact that the transmission loss of bottom-reflected sound is nearly independent of range has an important effect on the maximum ranges obtained with sonic gear. If the available signal output is between 60 and 80 db, the maximum range is likely to be less than 1,000 yards and unlikely to be greater than 2,000 yards. Contact is not established until the target becomes audible by way of direct sound. If the available signal output is greater than 80 db, however, the bottom-reflected sound may become useful, and range may suddenly increase to between 10,000 and 20,000 yards.

ULTRASONIC LISTENING

Ultrasonic sound is made audible by heterodyning, so that the loudspeaker of the listening system emits audible sound. The general principles of recognition for heterodyned ultrasonic

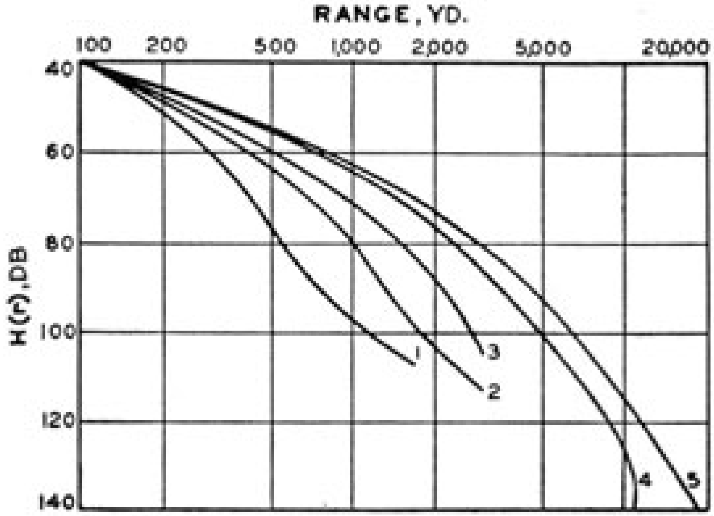


Figure 3-22 -Transmission loss *H*(*r*) at 24 kc for various thermal conditions.

sound are thus identical with those applying to audible sound. However, several quantitative differences exist.

In the first place, ultrasonic receivers usually have pass bands not more than 1 kc wide. The spectrum of the heterodyne output may thus be confined to the range of from 300 to 1,330 cycles per second, as compared with a range of 10,000 cycles per second in sonic listening.

In the second place, a 1-kc band of one ultrasonic spectrum is very similar to a 1-kc band of another. There are no single-frequency peaks, and although most spectra slope 5 to 9 db per octave, the change in spectrum level over a 1-kc band is negligible for many purposes. This principle applies to background noise as well as to the sound output of ships.

Thus, there usually is no one frequency of the heterodyned sound that is more audible than another. There is no tonal quality to distinguish the signal from the background.

In general, the recognition differential for ultrasonic listening is zero. The ultrasonic sound from a ship's screw, however, is usually rhythmically modulated

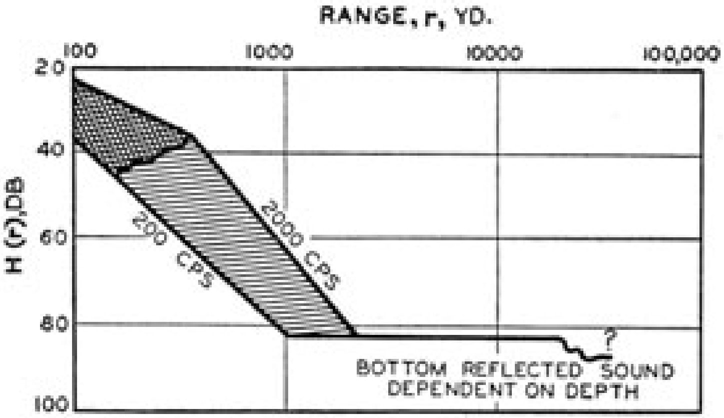


Figure 3-21 -Transmission loss *H*(*r*) for sonic sound.

in intensity. Recognition occurs when the maximum level of a rhythmic signal is equal to, or possibly a few decibels less than, the average level of nonrhythmic background. The maximum level of screw sounds is usually about 3 db above the average level. Because most measurements yield average values, they must be increased by about 3 db in calculating the available signal. This increase is sometimes loosely called a "recognition differential."

An exception to these statements occurs when the target vessel is echo ranging. The pings are heard as tonal pulses of sound which have a high recognition differential, as well as a high source level.

These considerations introduce some simplification into the calculation of ranges. The spectra of the signal and the background noise need not be considered in detail; it is sufficient to state the spectrum levels at the midpoint of the listening band.

The situation with regard to background noise is similar to that of sonic listening. That is, if the listening vessel is quiet, ambient noise predominates; whereas if the listening vessel is noisy, the noise of the listening vessel predominates. In ultrasonic listening, however, when ambient noise is limiting, shrimp crackle becomes important. The ordinary levels of ultrasonic ambient noise range from -78 to -53 db depending on sea state. If shrimp are present, however, the ambient noise levels may be -49 to -39 db.

When used at ultrasonic frequencies, listening gear discriminates against ambient noise. A directivity index, *D*, of -23 is common among

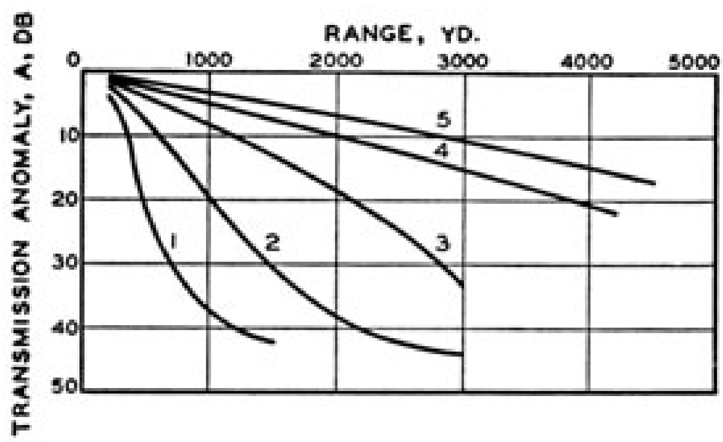


Figure 3-23 -Average transmission anomaly under various oceanographic conditions.

these curves (figure 3-22) the ultrasonic ranges should show less variation than do the sonic ranges. Because of this fact, also, there seems less probability of achieving great improvement in the performance of ultrasonic systems by a reduction in self-noise.

The values of the transmission anomaly in figure 3-23 have been determined by experiment. In this figure D_2 is depth for which the change in temperature is 0.3° *F*, which is the smallest temperature change that can be detected by the present bathythermograph. Note that D_2 has the following values:

standard echo-ranging transducers.

The graphs of figure 3-22 should be compared with figure 3-21 to contrast the transmission loss of the ultrasonic frequencies.

The curves in figure 3-22 are based on the anomaly of figure 3-23, and the same numbering is used. Because there is no horizontal portion of

Curve 1-0 ft $< D_2 <$ 5 ft.

Curve 2-5 ft $< D_2 <$ 20 ft.

Curve 3-20 ft $< D_2 <$ 40 ft.

Curve 4-40 ft $< D_2 <$ 80 ft.

Curve 5-80 ft $< D_2 <$ 300 ft.

Sonar Listening Systems

SONIC LISTENING SYSTEM

Many of the problems that affect underwater detection by receiving and analyzing both sonic and ultrasonic sound energy have been discussed in this chapter. Block diagrams and a brief description of the function of the various components of both types of listening equipment will now be given. The sonic listening equipment consists of a hydrophone, training unit, receiver-amplifier, and headphones or speaker. This system is shown in figure 3-24.

Hydrophone

The hydrophone used in listening equipment is primarily of the magnetostriction type. Some equipments use the crystal type. The primary purpose of the hydrophone is to convert sound energy in the water into electric energy that can be amplified and heard from the loudspeaker. The hydrophone must have a directional characteristic. This directional characteristic is used in two ways. First, it allows the operator to discriminate against unwanted sound, and, second, it enables the operator to determine the direction from which the desired sound is coming.

The discussion thus far applies to both sonic and ultrasonic hydrophones. There is little difference between them. The sonic hydrophone must have larger dimensions than the ultrasonic hydrophone for the same directivity index.

Training Unit

The method of training the hydrophone may be either manual or power. The more modern equipments use an electrically operated drive called a *servo-mechanism*. This drive is usually an amplidyne system. The operator can train the hydrophone on any bearing relative to the ship. A synchro repeater system is used to give bearing indications to the operator in both relative and true bearings. The two functions of the training device are therefore to allow the operator to train the hydrophone and to provide him with visual indication of the bearing to which the hydrophone is trained. The training unit is the same for both sonic and ultrasonic listening. In some installations it is used for both.

Receiver-Amplifier

The receiver-amplifier is simply an audio amplifier with wide-frequency response. The electrical signals from the hydrophone enter the receiver where they are amplified until their intensity is sufficient to drive a loudspeaker or headphones, as the need may be. There is no necessity for any frequency conversion because the signals entering the receiver in sonic listening are already in the audible frequency range-unlike those entering the receiver in ultrasonic listening in which heterodyning is necessary. This frequency conversion is the principle difference between the sonic and ultrasonic systems. It may be desirable under some conditions to limit the frequency response of the receiver in sonic listening. Band-pass filters are usually included for, this purpose. If the principal signal desired by the operator were about 500 cycles per second it would be possible to increase the signal-to-noise ratio by cutting down the band pass of the receiver just enough to include this frequency. Any noise falling outside the band pass of the receiver would not be heard.

Headphones and Speaker

The choice of headphones or speaker is dictated by the airborne noise of the surroundings. The purpose of these devices is to convert the electric signals from the receiver into sound impulses that can be heard by the operator. The cycle is completed by this conversion. In the water the signal is first sound energy, and when it falls on the hydrophone, electric impulses are generated. These electric impulses are amplified and reconverted into sound by loudspeakers or headphones.

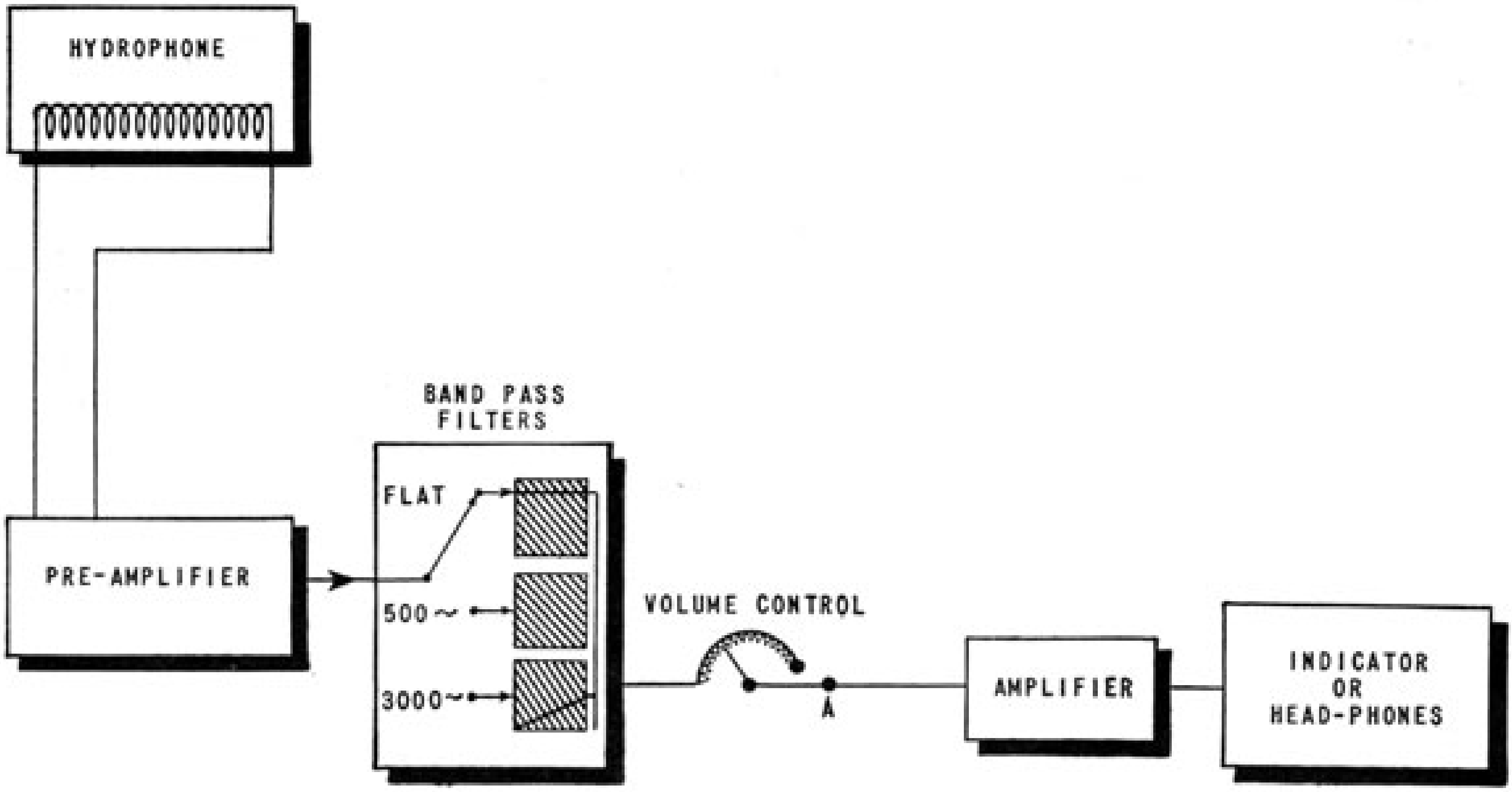


Figure 3-24. -Block diagram of a sonic listening equipment.

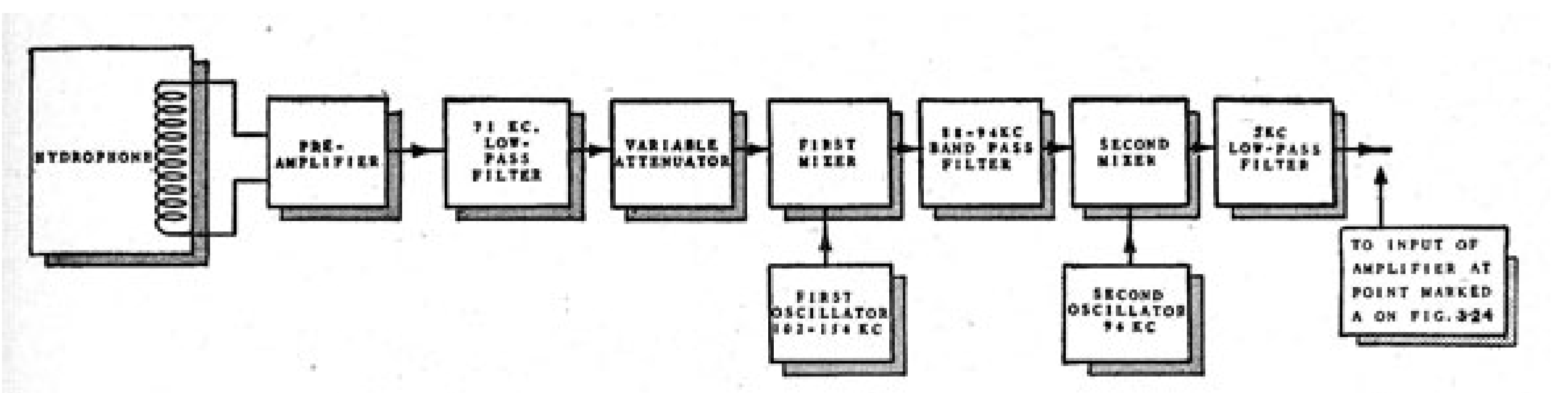


Figure 3-25 -Block diagram of the ultrasonic converter.

ULTRASONIC LISTENING SYSTEM

Ultrasonic listening equipment will be discussed by comparing it to sonic listening equipment. The ultrasonic hydrophone is the same as the sonic hydrophone in most cases. A small ultrasonic hydrophone, however, gives directivity similar to that of a large sonic hydrophone. Thus the directivity index of the same hydrophone used for both sonic and ultrasonic listening is greater for ultrasonic listening. An ultrasonic hydrophone, therefore, gives sharper bearing indication.

The training equipment of ultrasonic systems is identical to that of sonic systems.

The principal difference in sonic and ultrasonic listening is in the receiver-amplifier. When the sound to be heard is in the ultrasonic frequency range some method must be used to bring it into the audible, or sonic range. Heterodyning in the receiver accomplishes this change in frequency. Note the signal path in the block diagram of the ultrasonic converter shown in figure 3-25.

Usually there is a broad-band amplifier stage at the receiver input. This stage is followed by a filter system and an attenuator. The signal is then fed into the first mixer, where it is mixed with the output of a variable-frequency oscillator. The tuning of this oscillator provides for the adjustment of the receiver to various frequency inputs. The signal from the mixed stage is amplified through an intermediate-frequency amplifier similar to that of any superheterodyne radio receiver. This intermediate frequency is usually above the frequency of the ultrasonic signal and is the sum of the ultrasonic signal and the output of the oscillator. The intermediate frequency is then fed into a second mixer where it beats with a second oscillator to give an output in the audible-frequency range. This converting system is in addition to the regular audio amplification of the receiver, which drives the speaker.

Previous Chapter Sonar Home Page Next Chapter

CHAPTER 4

ESSENTIALS OF ECHO-RANGING EQUIPMENT

Basic Components of Echo-Ranging Systems

So far this textbook has dealt primarily with the behavior of sound in sea water, the ocean climate and characteristics, and marine life, all of which affect the transmission and reception of sound. Listening systems were described briefly at the end of chapter 3. The present chapter gives particular attention to echo-ranging systems and problems associated with echo ranging.

The typical echo-ranging system to be discussed here is of the directional-beam (searchlight) type. Particular emphasis is given to transducers because of their importance in the development of sonar. The purpose of this chapter is to give the reader a broad understanding of the basic components needed in an echo-ranging system.

TRANSMITTER

The source of the signal is called the transmitter and can be compared directly with the transmitter of a radio station. There is a variable-frequency oscillator that generates a signal in the frequency range of from 17 to 27 kc. This signal is fed into a wide-band amplifier that increases the signal intensity to about 400 watts if a magnetostriction transducer is to be driven, and to about 150 watts if a crystal transducer is to be driven. This signal is then used to drive the transducer. The length of transmission is controlled by keying contacts on the timing device. The measurement of time is started at the same time as the transmission. The timing device closes the keying relay at zero indication on the range scale. This action connects the transducer to the output of the transmitter and at the same time keys the oscillator for a in figure 4-1, which is a block diagram of the signal circuits.

SIGNAL USED IN ECHO RANGING

Signal Frequency

Practical considerations set rather definite upper and lower limits to the frequencies that can be used in echo ranging. The use of sonic frequencies (less than 10 kc) has not been considered practicable because of directivity requirements. A second reason for the use of ultrasonic sound is provided by considerations of the detectability of echoes. Echoes must always be detected against a background of interfering noises. Although these noises include sound of ultrasonic frequencies, the greater part of their energy is in the sonic region. Hence, ultrasonic echoes are masked less than sonic ones.

An upper limit to the practicable frequency is set by the attenuation of the sound in the sea. The attenuation coefficient increases very markedly with frequency. Hence, for search purposes, where long ranges are required, a frequency higher than about 25 to 30 kc is not suitable. When the range is being closed, and great accuracy of bearing is needed rather than a long range, the greater directivity associated with higher frequencies is the determining factor, and thus frequencies of from 50 to 100 kc may be useful. These frequencies ate especially useful for depth determination, where an extremely narrow beam is required; and because accurate depth determination is practicable only at comparatively short ranges, the high attenuation consequent to using high frequencies is not

predetermined length of time. When this time is over the relay returns the transducer to the receiver, so that any echo may be heard. The signal in the echo-ranging equipment may be followed significant.

The United States Navy at first adopted a compromise value of 24 kc. This frequency allowed fair directivity to be achieved while the size of

the transducer could be kept within practical limits. The attenuation was moderate. equipment using only 24-kc frequency is now being replaced by more elaborate equipment that can emit various frequencies, for the reasons just mentioned.

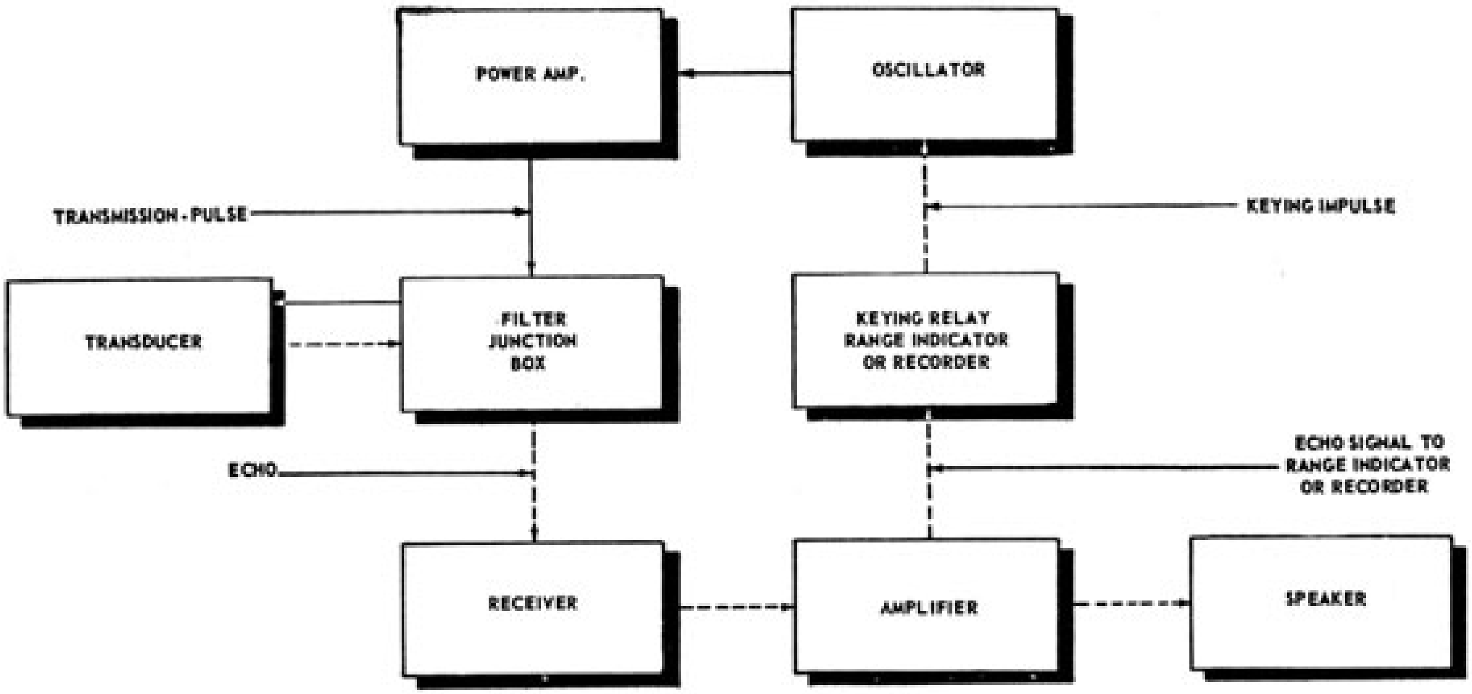


Figure 4-1 -Block diagram of an echo-ranging equipment.

Receiver-Amplifier

When the signal has been emitted, the transducer connections in the filter junction box are changed so that the transducer can act as a receiver of sound waves. An echo from a target or other sounds of proper frequency incident on the transducer plate produce oscillations that represent the various signals, but the frequency of these sounds is too great to be perceptible to the human ear. The mechanical pressure of the sound waves is converted into alternating currents by the magnetostrictive or piezoelectric effect. These signals must be amplified and changed to and signal frequencies. This intermediate frequency might be 60 kc.

The i-f currents which represent the signal are amplified further and utilized to render the incident sound perceptible in various ways. Note in figure 4-1 that the signal may leave the amplifier by two different paths. The customary methods of portrayal are:

1. The amplified i-f currents are heterodyned to sonic frequencies, which are converted to airborne

frequencies in the audible range, or they may be portrayed so as to be interpreted by the eye. The purpose of the receiver is to amplify the signal and present it in a suitable form.

The receiver is a superheterodyne type similar to the one described for ultrasonic listening equipment. The usual method is to amplify the signal at ultrasonic frequency, then mix it with a frequency from a local oscillator to obtain an intermediate frequency that is the sum of the oscillator

239276°-53-6

sound waves (made audible) by means of a loudspeaker or headphones.

2. The amplified signal voltage may be rectified and delivered to a "chemical range recorder," which utilizes the chemical effect of the current to record the range on a specially treated paper. The density of the trace is determined by the magnitude of the current. Thus the pulse of current representing the echo signal leaves a spot that is much denser than that part of the trace which represents the reverberation and noise. The range can be read from a scale opposite the

spot. An advantage of this method, not possessed by the others is that it provides a comparatively permanent record of incident sounds.

3. A range indicator used with earlier models of echo-ranging equipment has a rotating neon light which flashes when the signal voltage is applied to it. The range is read from an adjacent scale at the time of the flash.

4. The amplified voltage may be rectified and applied to a cathode-ray oscilloscope by the following methods:

a. The spot of this indicator is usually made to move along a vertical *y* axis to indicate range. The rectified voltage may be applied so as to cause the spot to deviate from straight-line motion (deflection in the direction of the *x* axis). The echo is then recognized by a greater *x*-axis deflection than that produced by reverberation.

b. In a second method of portrayal involving the use of a cathode-ray oscilloscope, the spot always moves in a straight line to indicate range. Its brightness is controlled by the rectified voltage from the receiver, so that the echo appears as a bright spot on the relatively dim, or invisible, line traced by the spot in the absence of an echo. This method is called *z*-axis portrayal.

It is possible to combine *x*- and *z*-axis portrayal.

Training Device and Bearing Indicator

Because the transducer is directional it is necessary to provide some method of rotating it in the horizontal plane, so that sound may be projected in any direction. The device for rotating the transducer is called the *training device*. The training device is power-operated by one of two methods-(1) an amplidyne system or (2) a thyratron system. Regardless of the method, the operator is able to control the position of the transducer.

To be able to train the transducer is not enough. The operator must be able also to determine the actual direction in which the transducer is trained and to read accurately either its true or its relative bearing. Bearing information is obtained from a device known as the *bearing indicator*.

Range Indicator

The timing device for measuring the time between transmission and echo is basically nothing more than an elaborate stop watch. It is called a *range indicator or range recorder* as the case may be.

The range indicator is a large motor-driven contactor that rotates a light behind a translucent scale. The scale is calibrated for various maximum ranges. When the light passes zero on the range scale the keying contacts are closed and the transmitter sends out a signal through the transducer. The transmission may last about one twenty-fifth of a second. When the keying contacts are broken the receiver is connected to the transducer and any echo received causes the rotating light to flash opposite the range corresponding to that from which the echo was received.

This procedure is exactly the same as that used in the first measurement of range when an observer held a stop watch, which he started when the transmission was made. He stopped it when he heard the echo. Because an ordinary stop watch gave very poor results, the British made a special one that speeded up the motion of the second hand. The United States Navy then improved on this method by developing the automatic type just described.

If the maximum range desired is 1,000 yards, the rotating light must make one revolution in the time required for the transmitted sound to travel 1,000 yards and return to the ship. If the velocity of sound is 4,800 feet per second, or 1,600 yards per second, the time for the sound to travel a total of 2,000 yards can be found from the equation

$r=vt$, or $t=r/v$, (4-1)

where r is the range in yards, v the velocity in yards per second, and t the time in seconds. If the values given in the example are substituted in equation (4-1),

t=2,000/16,000= 1.25 seconds.

The time for a 1,000-yard range is 1.25 seconds. This is a convenient time unit to remember because it is the time per thousand yards of range.

If the time for a 5,000-yard range is desired it is necessary only to multiply 1.25 by 5 and get 6.25 seconds.

The foregoing calculations indicate that the rotating light must make one revolution every 1.25 seconds for a maximum of 1,000 yards and that it must key the transmitter each time the light passes zero (once each revolution).

The range recorder was developed by the British still later than the range indicator and was used in World War II. The version used by the United States Navy was an improvement over the British model, but the principle was exactly the same. A recording paper was treated so that an electric current passing through it would make a mark on it. This paper was made to move at a uniform rate by a motor. This uniform rate of paper motion gave a time axis for the range plot. A stylus was caused to move perpendicular to the motion of the paper at a rate proportional to the echo-ranging velocity of sound, or 800 yards per second. The motion of the stylus gave a time-range plot of the echoes. When received, the echo causes a current to pass from the stylus through the paper and leave a mark on it opposite a range scale placed over the paper. The range recorder has the advantage of "memory" over the range indicator. With the range indicator, the flash of light is gone once it is made, and if the observer misses it he cannot get a second look. The recorder, however, gives a permanent record of each echo.

The recorder also operates the keying relay. The transmitter is keyed just as the stylus of the recorder starts to move across the paper. The position of the stylus at any time is then proportional to the range.

Transducers

It has been pointed out in chapter 1 that a vibrating body with dimensions that are small compared to the wavelength of the sound, radiates sound energy in all directions. If, however, the dimensions are large compared to the wavelength of the sound, the propagation becomes directional. Radiation from emitters of the first type is called *spherical radiation*; that from emitters of the second type is called *directional radiation*.

In underwater sound, the important consideration is the production of directional transducers. It is important to know the direction from which a sound is heard or received. In some systems, the sound emitted is from a cylindrical source and is transmitted horizontally in all directions at the same time. The vertical dimension of the beam, however, is made narrow and sharp. Even with this type of transmitter, it is important to receive with a directional transducer. energy in the water into electric impulses in the receiver system.

ULTRASONIC SOUND SOURCES

Sound waves having frequencies above the audible range have been produced and used in laboratories for many years, but only in relatively recent years has it been possible to generate such waves of sufficient energy for practical use. The most common method of producing ultrasonic sound at the present time is by causing formed bodies to vibrate at a high natural frequency (resonance) by the application of rapidly oscillating electric voltages.

Magnetostriction Effect

The change in length of a rod or tube of ferromagnetic material when it is placed in a

In sonar a transducer is a device which may convert electric energy into acoustic energy in the surrounding water. Such a transducer may also convert acoustic energy from the surrounding water into electric energy. It is both a transmitting and a receiving device. The term "projector" is commonly used for the device the specific function of which is to transmit sound energy into the water. The *hydrophone* is a device used specifically for converting sound magnetic field parallel to its length is called *magnetostriction*. Nickel, annealed cobalt, and a few alloys of nickel possess a more pronounced magnetostriction effect than other metals.

The phenomenon is not related in any simple manner to other magnetic properties. Figure 4-2, A, shows the relative change in length dL/L, as a function of field strength in gausses, for several materials. The change in length, although small,

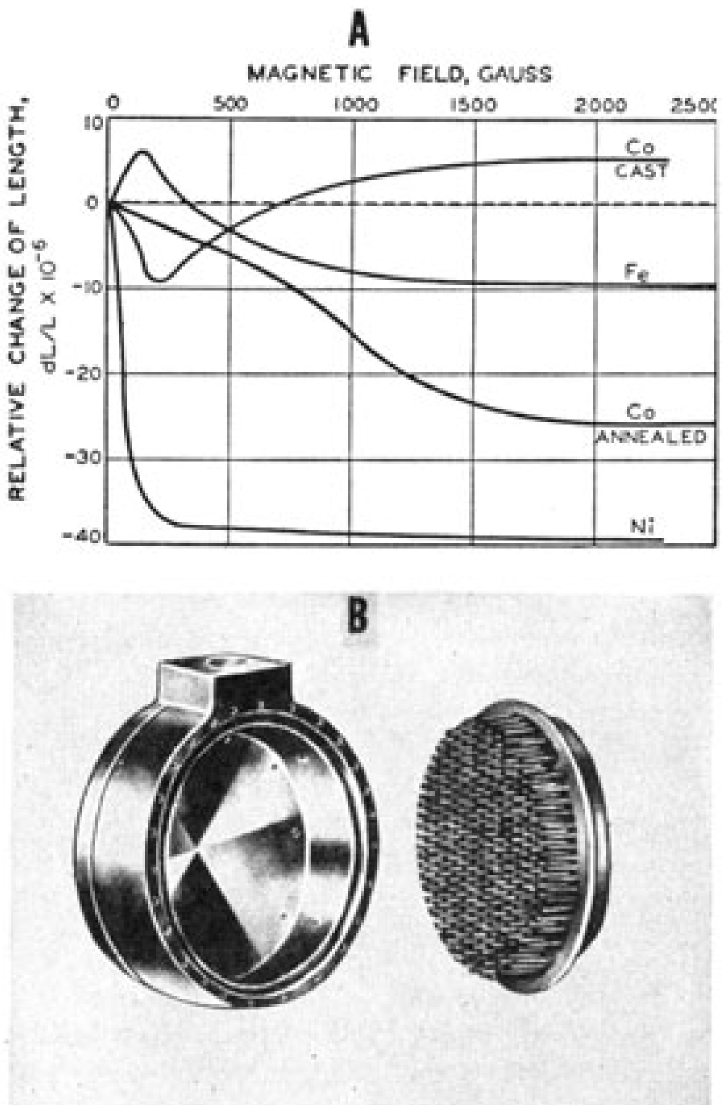


Figure 4-2 -Magnetostriction transducer. A, Magnetostriction in iron, nickel, and cobalt; B, construction of a magnetostriction transducer head.

Magnetostriction is reversible. If a previously magnetized rod of nickel is stretched, the magnetization of the rod is decreased; if it is compressed (in the direction of its length), the magnetization is increased.

Magnetostriction Sound Sources

Magnetostriction becomes a source of sound waves when a nickel rod is subjected to an alternating magnetic field by winding a coil of wire around it and sending an alternating current through the coil. The rod is shortened periodically in response to the changing field. Because the shortening of length is independent of the direction of the field, the rod is shortened when the current goes through the positive half of its cycle, regains its length as the current becomes zero, and is shortened again when the current goes through the negative half of its cycle. Thus, the rods goes through two cycles of motion while the current completes one oscillation. The doubling of the rod vibration frequency can be prevented by subjecting the rod to a constant magnetic field with properly arranged permanent magnets or by sending a constant direct current (polarizing current) through the coil. Figure 4-2, A, shows that if a nickel rod is initially shortened to some point on the steep portion of the curve by placing it in a constant magnetic field and is further subjected to the magnetic strains imposed by an

depends upon the strength of the magnetic field, but is independent of the direction of the field. In addition, its magnitude depends on (1) the material, (2) its heat treatment and present temperature, and (3) the degree to which it was previously magnetized.

Figure 4-2, A, shows that nickel possesses the property of magnetostriction to a much greater degree than any other metal. It decreases in a fairly linear manner for an increasing field strength up to about 200 gausses. If the field is increased beyond this value, the additional change becomes extremely small. The maximum relative change in length is about 40 parts in a million. However, because Young's modulus for nickel is high (30 X 10^6 lb/in^2), a large force is exerted against anything that resists this small change in length.

alternating current, it can be made to shorten and lengthen in step with the alternating current. The polarizing magnetic field not only prevents a doubling of the frequency of vibration of the rod, but also allows operation on the steeper and more linear portion of the curve. This characteristic is an important advantage.

The natural fundamental frequency of vibration, *F*, of a rod of length *L* is

$$F=(\tfrac{1}{2}L)\,(M/\sigma)^{\frac{1}{2}} \quad (4\text{-}2)$$

where *M* is the modulus of elasticity and σ is the density of the material. If a current of this frequency is sent through the coil, the amplitude of oscillation is a maximum; relative changes in length may be of the order of 1 in 10,000. Calculations using equation (4-2) show that a rod of nickel 5 inches long has a fundamental frequency of vibration of about 20 kc; and that one 1.6 inches long resonates at 60 kc.

If a nickel rod is set in vibration in the manner just described, sound waves, with a frequency determined by the frequency of the magnetizing current, are emitted from the end of the rod. To obtain the maximum possible intensity, a practical transducer is constructed by embedding the end of several hundred small nickel rods into a steel diaphragm of dimensions which ensure that its resonance frequency is the same as that of the rods. Each rod is excited by its own coil.

A typical magnetostriction transducer is shown in figure 4-2, B.

Because of the reversibility of the magnetostriction effect, the transducer acts also as a receiver. Sound waves impinging on the diaphragm compress or extend the rods; corresponding changes in the magnetization of the

rods induce alternating currents in the coils, which after amplification can activate a portrayal device.

Piezoelectric Effect

When subjected to a mechanical stress, some crystals-such as quartz, Rochelle salt (RS), ammonium dihydrogen phosphate (ADP)- exhibit electric charges on certain surfaces. This phenomenon, called the *piezoelectric effect*, was discovered by the Curie brothers in 1880. The electric charges developed are proportional to the stress applied to the crystal, and the charges are of opposite sign for compressions and tensions. Shortly after this discovery, the Curies found the inverse effect to be equally true-that is, when a

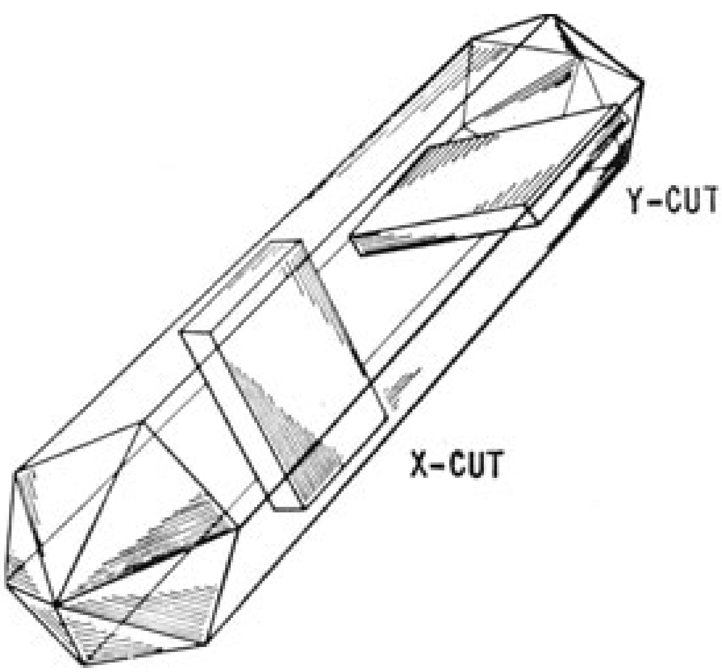


Figure 4-3 -Quartz crystal, showing X-cut and Y-cut plates.

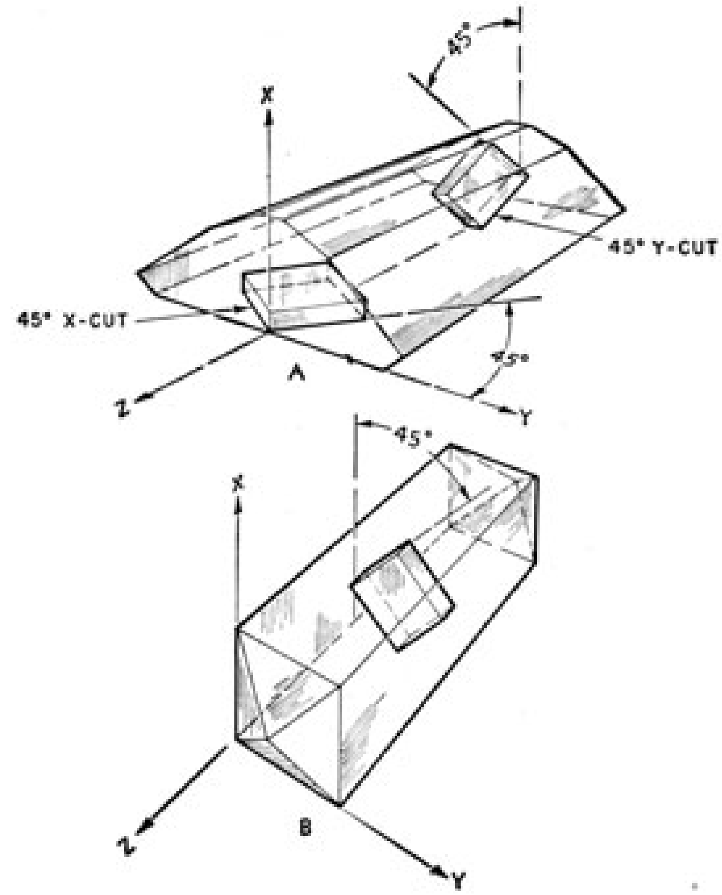


Figure 4-4 -Crystals, showing orientation of rectangular plates for: A, 45° X-cut and Y-cut RS (top) and B, 45° Z-cut ADP (bottom).

crystal is subjected to an electric field, mechanical strains occur in the crystal. Thus, these two effects are exactly reversible and a direct proportionality exists between cause and effect, in both magnitude and sign. The fact that magnetostriction is a nonlinear effect, where as the piezoelectric effect is linear, serves as an important distinction between these two phenomena.

If a piezoelectric crystal is placed between two electrodes and an oscillating electric voltage is applied to the electrodes, the crystal vibrates. Because the elastic properties of such crystals differ in different directions, the vibrations occur in different ways, depending on the orientation of the crystal relative to the electrodes. In any case, the natural frequency of vibration is given by an equation similar to equation (4-2), where the value of the elasticity modulus differs for different

orientations of the crystal.

A crystal as used in this book indicates a properly oriented piece cut from the mother bar. If such a crystal is equipped with suitable electrodes and properly mounted and protected it serves to generate or receive sound signals.

Rectangular plates cut from the mother bar at various angles of orientation are shown for quartz (figure 4-3), RS (figure 4-4, A), and ADP (figure 4-4, B).

Those crystals designated as 45° X-cut and 45° Y-cut RS and 45° Z-cut ADP are the only types of cut crystals that have so far found extensive practical application in underwater sound transducers in the United States. The use of 45° X-cut RS is now limited to special and rare circumstances, such as for small hydrophones on long cables where a preamplifier cannot be used. The use of 45° Y-cut RS has declined greatly. On the whole 45° Z-cut ADP is preferable unless some particular reason (such as low frequency) indicates otherwise. Quartz has been used effectively in England, but only because of an inadequate supply of RS and ADP crystals.

Quartz has the advantage of being strong and insoluble in water, whereas RS and ADP are fragile and soluble. Solubility is a disadvantage in all seagoing applications, although it can be overcome by suitable precautions in the design and construction of transducers. In the laboratory, on the other hand, solubility is an advantage in that it makes possible the production of good artificial crystals, whereas quartz must be mined, and only a small fraction of quartz crystals found are large enough and perfect enough for acoustic purposes. Quartz also has the disadvantage of being very hard and consequently difficult to cut and polish. Both RS and ADP crystals are soft enough to be cut with a band saw and shaped by ordinary metal-working power tools, if care is exercised to prevent chipping.

The British Asdic, the forerunner of our sonar, utilized X-cut quartz crystals. These crystals were laid flat on a steel plate, as shown in figure 4-5, A, and arranged in a mosaic so that the plate was adequately covered. An identical plate (not shown in the figure) was then laid on top of the crystals, thus forming a sandwich. The assembly was made mechanically rigid by means of clamps at the edges of the plates. Insulating washers made it possible to connect the plates to the terminals of the a-c source.

The deformation of the crystal when the voltage is applied is shown in figure 4-5, A, by the arrows. When the potential of the upper face is positive, the thickness increases. Simultaneously, the other two dimensions shrink. The changes which occur in the length, width, and thickness are such that the volume of the plate remains the same.

When the potential is reversed, the deformations are in the opposite direction. The two faces are not equivalent; hence, care must be taken to arrange all

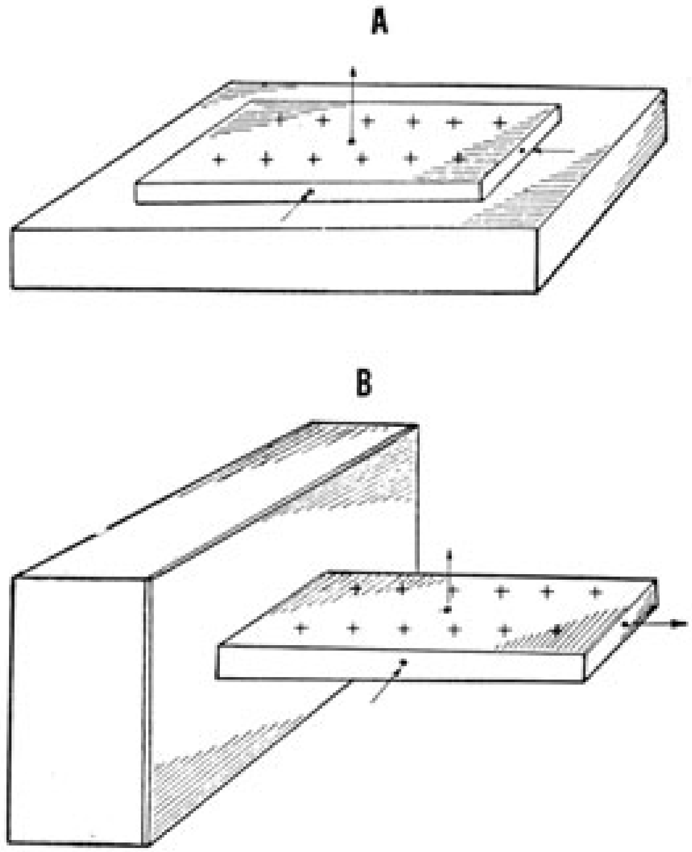


Figure 4-5 -Mounting of transducer crystals. A, Asdic transducer; B, RS and ADP crystals.

the plates in a mosaic so that they expand and contract "in step." Because the plate is compressed during one-half of the cycle of the a-c field and extended the same amount during the other half, it vibrates with the same period as that of the field. If this is the natural frequency of the crystal, the amplitude of vibration is a maximum. The natural frequency of the thickness vibrations, the one used in the Asdic transducer, calculated from equation (4-2) is

F=285.5/t kc, (4-3)

where t is the thickness of the plate in centimeters. However, experiments showed that this relation is only approximately true, because the plates generally execute vibrations in other modes than the ones mentioned; moreover, besides compressional vibrations, vibration due to shear may

deformation resulting from the indicated charge. There are many designs and methods used in assembling crystals into a transducer; however, in general, the crystals are cemented to a single heavy plate. To prevent short-circuiting, the surface of the backing plate must be enameled.

Many crystals are mounted on a single backing plate, as shown in figure 4-6, and sound is radiated from the free ends of the crystals. They are protected from the sea water by a "window." The window may serve to separate two liquid media, as sea water and castor oil, or the crystals may be attached directly to the inside of the window, the window not only protecting the crystals from the action of the sea water but also serving as a means of support. The space not occupied by crystals, between the backing plate and window is filled with carefully purified castor oil. Rubber has been widely adopted for

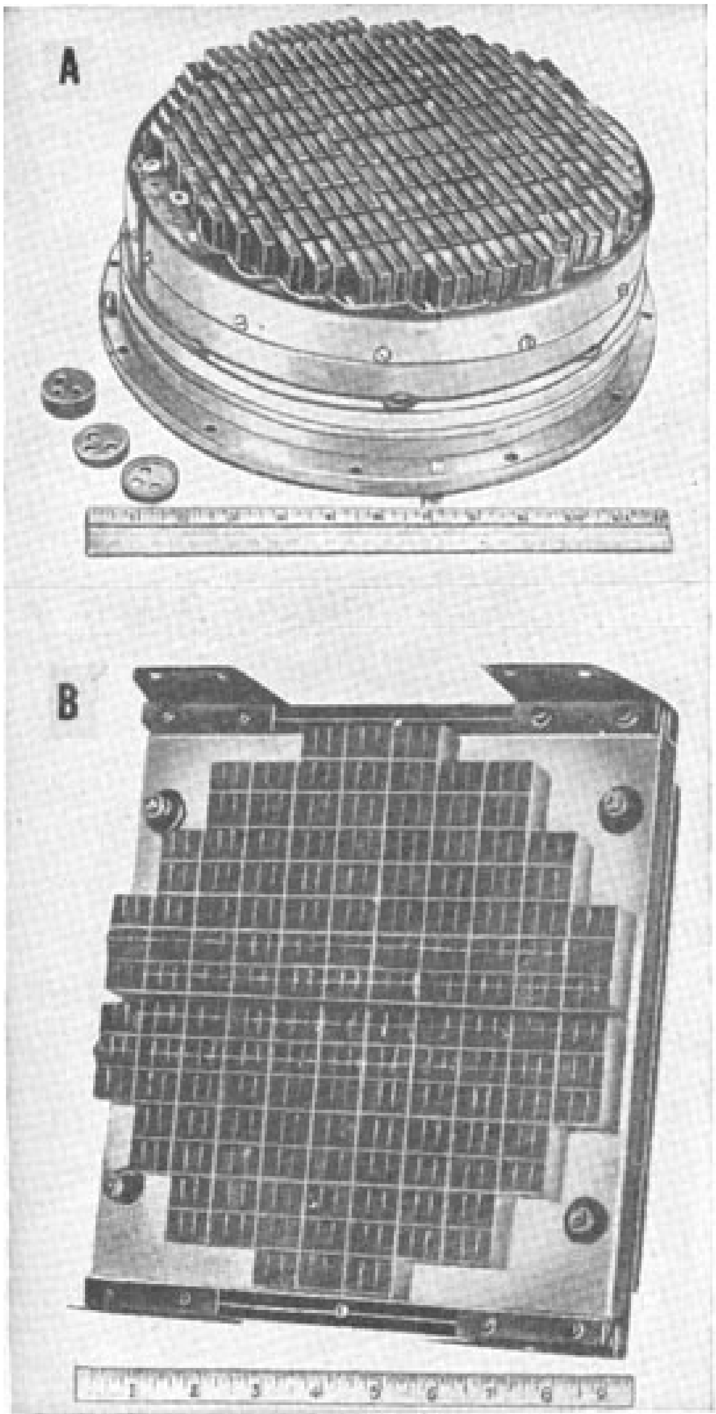


Figure 4-6 -Types of crystal stacks for transducers.

also be present. Such vibrations, coupled to the primary ones, tend to change the primary frequency of vibration.

Rochelle Salt and ADP Transducers

Sonar transducers using plates of Rochelle salt and ADP crystals are mounted so as to utilize the acoustic windows in crystal transducers, primarily because of the good impedance match obtainable with sea water but in part due to its elastic properties, abrasion resistance, and its electric resistivity.

The resonant frequency of the length vibrations of the crystal plates, as shown in figure 4-5, is a function of both the length, *L*, and the width, *w*, of the plate; it is generally multiplied by the length to form a term called the *frequency constant*-

FL=64.7-(13.6) $(w/L)^2$ kc. (4-4)

DIRECTIVITY

One important property of a transducer, when acting as a projector, is the manner in which transmitted energy is distributed in direction; or when acting as a receiver, the dependence of its sensitivity on the direction of the incident sound. energy.

Let us now look into some of the principles involved in the design of a directional transducer. Brief consideration has already been given to the directional properties of a large source-one with linear dimensions several times as great as the wavelength of the emitted sound, as compared to the omnidirectional properties of a "point" source. Directional transmission of sound results from the interference of waves spreading out from two or more poi at sources or from several points on a large source.

length vibrations instead of the thickness vibrations, as shown in figure 4-5, B. The two large faces are coated with a metal foil, and the a-c voltage is applied to the foil. The arrows indicate the

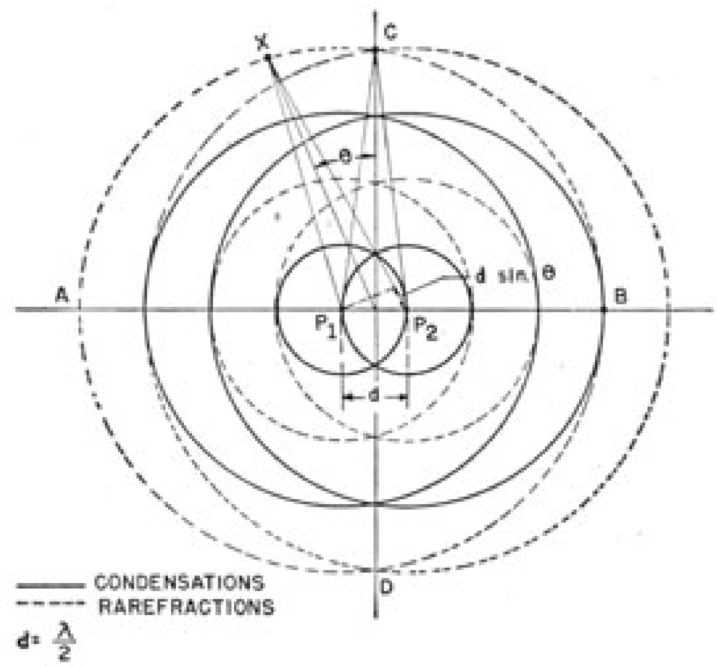


Figure 4-7 -Interference of waves from two sources, for *d*=λ/2

Consider two point sound sources, P_1 and P_2, in figure 4-7, located a distance *d* apart equal to one-half wavelength, vibrating in phase with the same frequency and amplitude. Along *CD*, the perpendicular bisector of the line that joins the two points, condensations from the sources arrive at *C* at the same time, as do rarefactions, and the interference is constructive. Thus the sound pressure at *C* is the sum of the pressures from each source. The transmitted sound energy is a maximum along line *CD*. At point *B*, on the line joining the two points, each source again exerts a pressure. In this case condensations produced by one source arrive with rarefactions due to the other and destructive interference results. The sound pressure at *B* is also the sum of the pressure from each source; however, because the waves from the two sources are arriving at *B* in phase opposition, the sound pressure at *B* is zero.

This special case of two point sources located one-half wavelength apart constitutes the basis for a directional transducer, with a maximum output along the perpendicular bisector of the line joining the two sources and zero output along the line joining the two sources. In directions between *AB* and *CD*, the sound pressure resulting from the combined waves varies with direction. The polar diagram, *B*, of figure 4-8, shows a complete picture of the distribution of sound pressure resulting from the interference of waves from two point sources spaced one-half wavelength apart.

If the two sources are separated by some other fraction of a wavelength, the difference in the pressures at points *C* and *B* depends on the amount of the separation. For example, if the separation is λ/10, the difference in pressure at *C* and *B* is about 5 percent. The smaller the separation of the sources-that is, the smaller the dimensions of the whole source relative to the wavelength-the smaller is the difference in pressure between points on the two lines under discussion.

If the point, *X*, under observation lies in a direction making an angle θ with the perpendicular bisector of the line joining the two sources (figure 4-7), the wave from one source lags behind the one from the other by a distance, $d \sin \theta$, where d is the distance between the two sources. The phase lag between the two waves arriving at the point *X* is $(2\pi d / \lambda) \sin \theta$ radians. The ratio of the resultant pressure, *p*, at

point X to the pressure,

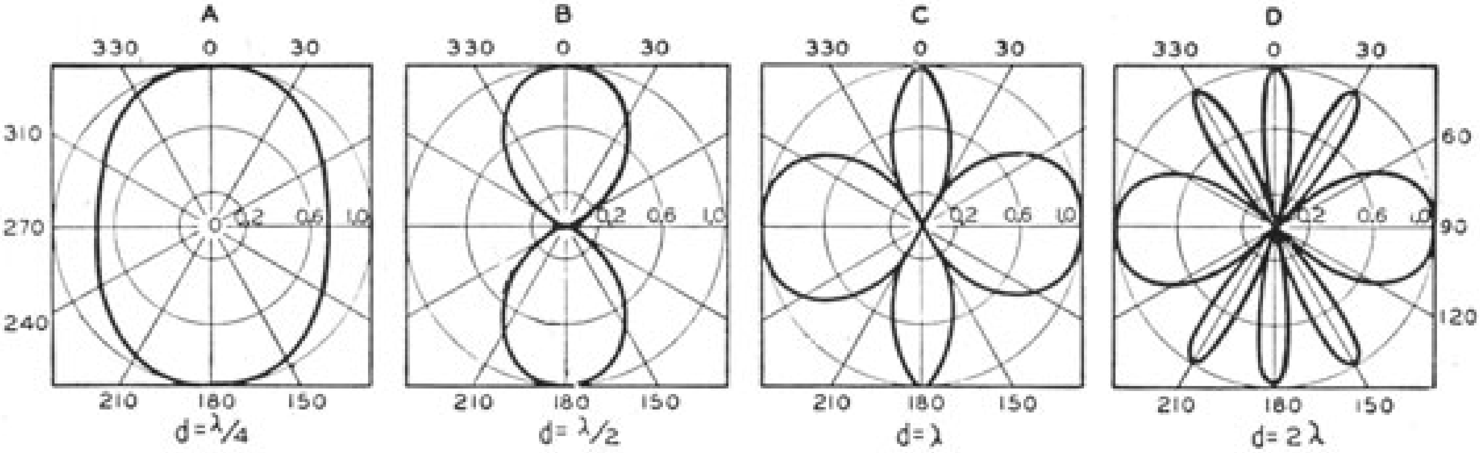


Figure 4-8. -Graphs of equation (4-5) for various values of *d*.

p_o, at the corresponding point, C, on the normal (θ=0) may be obtained by vector addition and is

$p/p_o = \cos(\pi d / \lambda \sin \theta)$. (4-5)

Graphs of this function (figure 4-8) normalized to a maximum value of unity show a series of maxima and minima for four values of *d* as θ is made to vary through 360°.

Practical sources of sound can be considered to be composed of a number of point sources. By reasoning similar to that just used, the pressure at any point in the field surrounding the source can be calculated. The calculations become extremely complicated for all but the simplest possible arrangements; however, they have been made for several simple geometrical configurations and are found in standard works on sound. A brief discussion and a few equations will be given to illustrate the problem involved.

There are three types of wavefronts that can be handled rather simply-that is, waves that are (1) plane, (2) cylindrical, and (3) spherical. Because the spherical wave represents a point source and

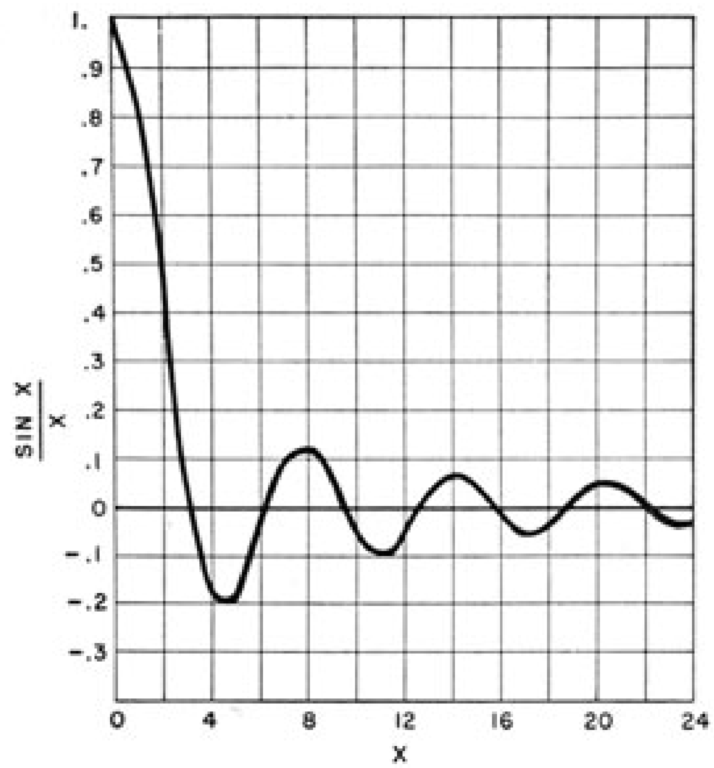


Figure 4-9. -Sin(*x*)/*x* as a function of *x* where *x*= (π*d*/λ)sinθ (square plane radiator).

The function that represents the square or rectangular radiator is

$p(x)=(ab/v_1)$ (sin x/x), (4-6)

where $x = ka/2 \sin \theta$ in the plane perpendicular to the

gives a nondirective field, it is of little use in transducer design. The plane and cylindrical cases, however, are useful in that most transducers have either plane or cylindrical radiating faces.

Mathematical calculations show that the sound field of any shaped-plane radiator should, close to the surface, exhibit many maxima and minima distributed in space both along the axis perpendicular to the surface and in planes parallel to the surface. Furthermore, such a sound field should at great distances, exhibit a central maximum with side lobes of decreasing amplitude with distance from the central axis. The field at a great distance is of course important in echo ranging, and the field close in is important in the coupling between two or more transducers that must be operated close together.

Most plane radiators in use are bounded by squares or circles and the chief interest is in the distant part of their sound fields. Under these conditions, the directivities are quite easily calculated. Although a mathematical analysis is beyond the scope of this book, a simple statement of the types of functions that represent the variation of pressure with the angle, θ, is of interest.

side a, or $x=kb/2 \sin \theta$ in the plane perpendicular to the side b. If only the variation with angle θ is needed the first term, ab/v_1 may be omitted because $(\sin x)/x=1$ when $x=0$. A graph of the function $(\sin x)/x$ normalized to unity, is shown in figure 4-9.

The case of the circular radiator was first solved by Rayleigh. A mathematical expression of the circular case involves Bessel functions and is not given here. The graphical representation, however, is very similar to the square radiator pattern shown in figure 4-9.

The zeros of the $(\sin x)/x$ function corresponding to the nulls between lobes come at $x=N\pi$, N=1, 2, ..., while the maxima, side lobe peaks, occur at x=4.5, 7.7, 10.9, 14.1, 17.1, 20.3, ...

Other useful facts about these functions are given in table 6.

Using the $(\sin x)/x$ function for the case of a plane radiator bounded by a square, the expression for p/p_o becomes

$$p/p_o = \sin(ka/2 \sin \theta) / (ka/2 \sin \theta), \quad (4\text{-}7)$$

where k is $2\pi/\lambda$, and a is the length of the side of the square.

with step variation over their surfaces. however, the continuously variable velocity method has given patterns for a wide variety of distributions which are a valuable guide for design and which also give a perspective to the lobe-suppression problem.

Several cases using two velocity distributions have been calculated for both circular and square surfaces. Experimentally, the velocity ratio of 3

TABLE 6 - *Useful Facts About Radiation Function*

If values of θ are substituted in equation (4-7), the pressure in all directions relative to pressure on the normal to the surface can be plotted for arbitrary values of *a* and λ. A directivity pattern is obtained if the results are plotted on polar coordinate paper.

Db down	Formula
	Circular Radiator α=radius of circular radiator λ=wavelength
3	$\theta=\sin^{-1} 0.258\ \lambda/\alpha$
6	$\theta=\sin^{-1} 0.305\ \lambda/\alpha$
∞	$\theta=\sin^{-1} 0.595\ \lambda/\alpha$ 1st zero
17.8	$\theta=\sin^{-1} 0.818\ \lambda/\alpha$ max first lobe
∞	$\theta=\sin^{-1} 1.111\ \lambda/\alpha$ 2nd zero
23.8	$\theta=\sin^{-1} 1.34\ \lambda/\alpha$ max 2nd lobe
∞	$\theta=\sin^{-1} 1.62\ \lambda/\alpha$3rd zero
	Square Radiator α=side of square radiator
3	$\theta=\sin^{-1} 0.446\ \lambda/\alpha$
6	$\theta=\sin^{-1} 0.605\ \lambda/\alpha$
∞	$\theta=\sin^{-1} 1.00\ \lambda/\alpha$ 1st zero
13.47	$\theta=\sin^{-1} 1.43\ \lambda/\alpha$ max first lobe
∞	$\theta=\sin^{-1} 2.00\ \lambda/\alpha$ 2nd zero
18.24	$\theta=\sin^{-1} 2.36\ \lambda/\alpha$ max 2nd lobe
∞	$\theta=\sin^{-1} 3.00\ \lambda/\alpha$ 3rd zero

To achieve a higher degree of directivity, the linear dimensions of the transducer must be several times as great as the wavelength of the sound energy. Sound of 10 kc in sea water has a wavelength of about 6 inches. To get a minimum degree of directivity at that or a lower frequency obviously would require a larger transducer surface than is practicable.

Directivity Patterns

It is customary to plot the directivity function *B*, or-10 log *b* (θ), rather than *b*(θ) itself; but this means the importance of the side lobes is stressed, as can be seen from figures 1-4 and 1-5. In echo ranging, the side lobes are important because an echo may be received along one of them and considered to be due to the sound of the main beam. Such a misinterpretation would result in a large bearing error. Thus the suppression of side lobes plays an important part in the design of transducers. For example, if the velocity of vibration over the surface of a plane transducer is not constant, but is less around the edges of the transducer than in the center, the side lobes are always reduced in magnitude. However, the main lobe is generally broader. Several methods of calculating the sound field from transducers of variable surface velocities and phases have been used.

For practical reasons transducers are not designed with velocities continuously variable but

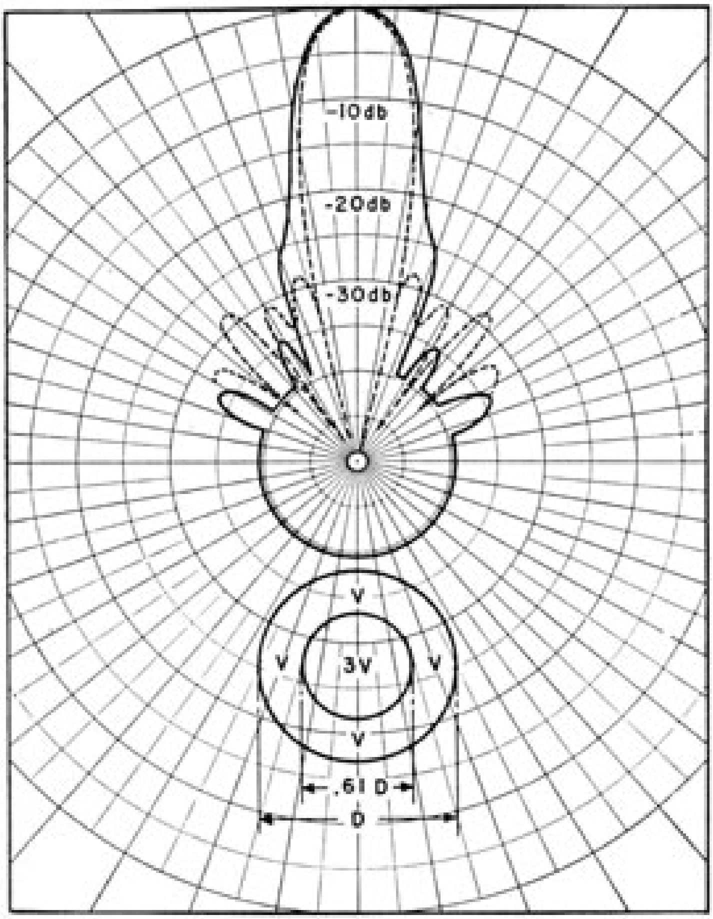


Figure 4-10 -Theoretical and experimental directivity patterns of a crystal transducer.

to 1 and a diameter ratio of 0.6 to 1 have given the greatest suppression of side lobes so far in the circular type, as shown in figure 4-10, which includes the theoretical pattern (broken line) as well as the experimental pattern (solid line). The relatively small size of transducers usually limits the number of velocity steps to two or three.

Directivities are usually calculated in some plane which is normal to the face of the transducer; and because the beam is three-dimensional, the plane in which a directivity pattern is measured must be specified. If the transducer is a circular type, the beam may have symmetry about the normal to the transducer face, as shown in figure 4-11, where the frequency is 25 kc and the diameter is 15

surface is uniform in velocity, the phases in adjacent lobes differ by 180°. (See figure 4-12, A.) By a reciprocal theorem, if the radiating surface is divided into zones the amplitudes of vibration of which decrease in magnitude and alternate 180° in phase in a manner similar to the lobe pattern of a uniform radiator, the pattern should be uniform over a certain arc and have no side lobes. Such a pattern is shown in figure 4-12 B. If a linear phase shift across the radiating surface is used, the main lobe is shifted in direction as shown by figure 4-13, in which the phase is shifted 30° per point radiator. Phasing of this type can be used to train the main lobe electrically while the transducer is fixed.

Theoretically it is possible to fashion the directivity of a transducer into any desired form. Success in such fashioning, however, requires the radiating surfaces to perform according to

inches. However, if the transducer is nonsymmetrical, there exists a directivity pattern for each possible axis of rotation, and in general these various patterns are different.

The effects of variations in surface velocities have been discussed. Phase variations also are important and both phase and velocity variations may be used simultaneously. If the radiating

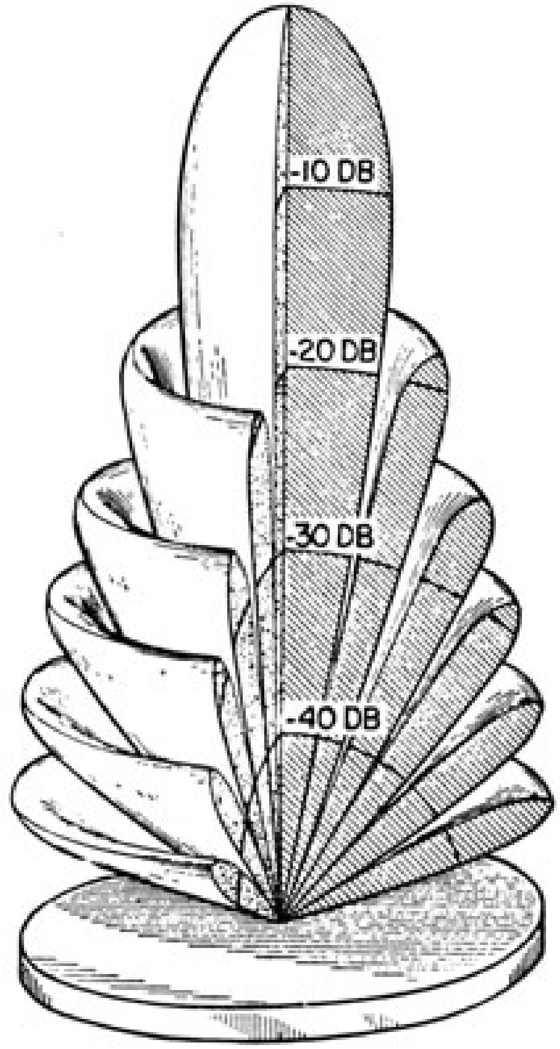
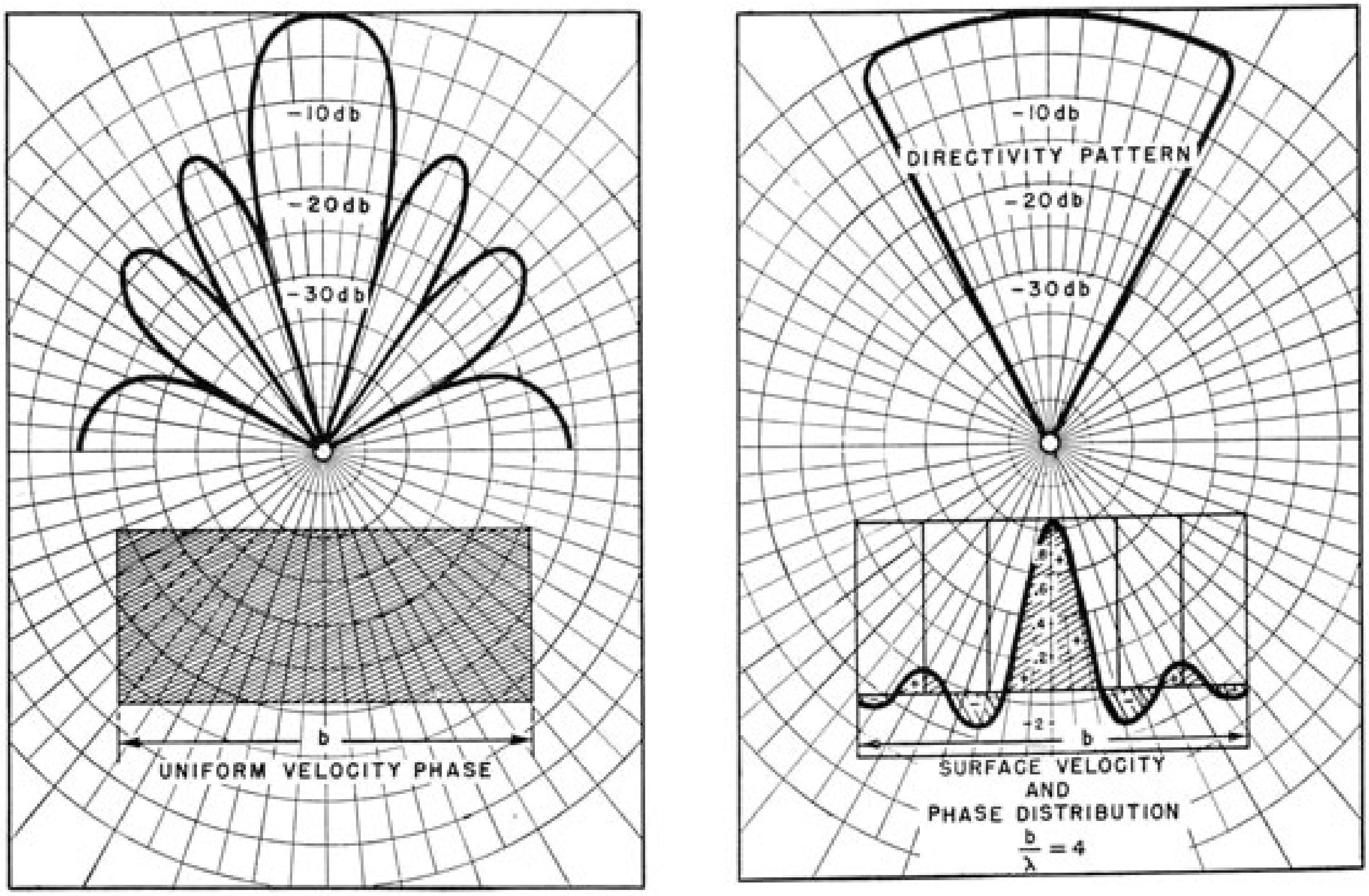


Figure 4-11 -Three-dimensional directivity pattern for a circular plate.



A B

Figure 4-12 -Reciprocal relation between the surface velocity function and the corresponding directivity function in a square-plane radiator. A, Uniform velocity in phase; B, surface velocity and phase distribution.

prescribed conditions, leading to one of the most difficult problems in the construction of transducers. Wide variations in the agreement between theory and experiment are encountered in transducers of different design, and often are encountered in particular units at different frequencies. These departures from theory vary in magnitude all the way from negligible departures to those large enough to render the unit useless for its intended purpose. The analysis of these eccentricities can be divided into two parts, one treating the main, or central, lobe and the other treating the side lobes. The most important feature of the main lobe aside from its absolute intensity is its width, which can be defined by two points on each side of the center that are 6 db down in intensity from the maximum. These theoretical beam widths for the square- and circular-plane radiators with this definition are

$$\theta=2 \sin^{-1} 0.605 \lambda/\alpha \text{ (square)}$$

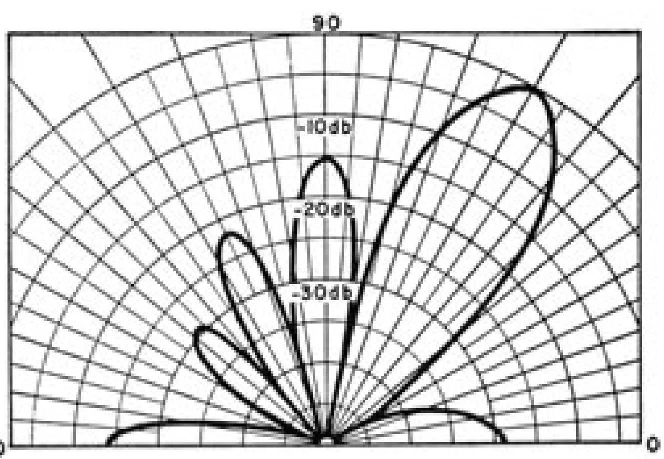


Figure 4-13 -Shifting of the main lobe by a

linear phase variation over the length of a line of point radiators.

and

$\theta = 2 \sin^{-1} 0.305\ \lambda/\alpha$ (circular).

(See table 6.)

In general the experimental beam widths are in good agreement with theory even when the accompanying side lobes are in very poor agreement. The width of the central lobe can thus, usually with good approximation, be predicted from the over-all dimensions of the transducer.

Detecting submarines under various conditions establishes requirements for echo ranging that can be met only by using several transducers. For general long-range search purposes, it is desirable to have a relatively wide beam with circular symmetry and small attenuation. For this purpose a circular transducer driven at 15 kc is suitable. For close ranges, a narrower beam can be achieved by using a transducer driven at 30 kc; the loss in range due to increased attenuation at the higher frequency is compensated for by the greater concentration of the beam and the greater accuracy in obtaining bearings on a target.

The QGA echo-ranging equipment is designed along these lines. The two transducers are mounted in a single dome, although they are operated independently. The system consists of two complete equipments, which are practically identical except that one operates at 15 kc and the other at 30 kc. Both transducers may be trained through 360° in azimuth. The 30-kc transducer may be tilted to 45° for maintaining contact with submarines that approach close enough to pass under the horizontal beam. The directivity patterns for the two frequencies of the QGA are shown in figure 4-14. The solid curve is the pattern for the 15-kc transducer; the dotted curve that for the 30-

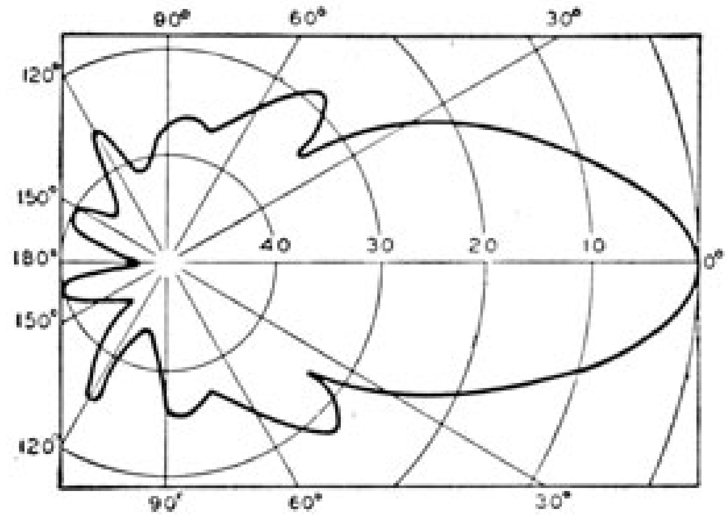


Figure 4-15. -Directivity pattern of magnetostriction 24-kc echo-ranging transducer.

on the axis indicate db below the maximum. The directional characteristics of the transducer are described by the directivity index which is a measure of the fraction of the sound energy that is sent out in the desired direction. The directivity index is expressed by a negative number. The larger the number numerically the more directional the transducer. The directivity index is described in more detail later in this chapter. The directivity index at 15 kc is -18.1 db; at 30 kc, -23.2 db.

Directivity patterns of transducers used in some of the older sonar equipments are shown in figures 4-15 to 4-18, inclusive. Figure 4-15 shows the pattern of the standard QC transducer, which consists of 608 hollow nickel tubes arranged on a circular diaphragm. Numbers on the axis indicate decibels below maximum. In this gear the tubes are arranged in circular form, and are pre-magnetized by a polarizing current. The directivity index is -21.4 db.

kc transducer. The numbers

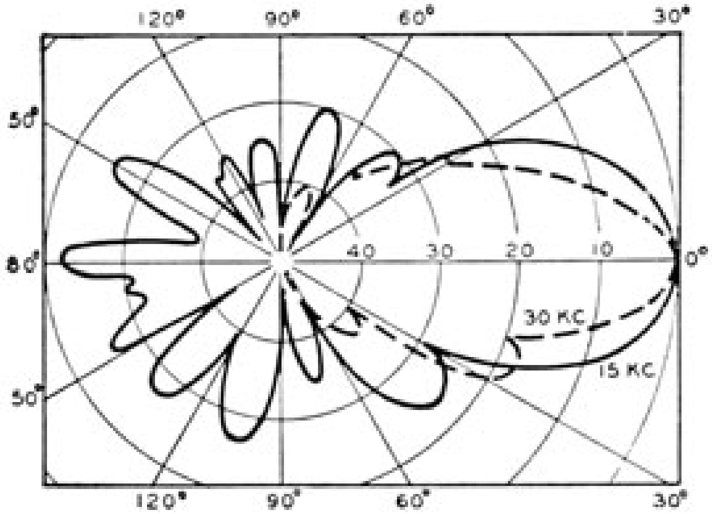


Figure 4-14 -Directivity patterns of QGA echo-ranging transducer.

Another form of QC gear, the QCU, has the directivity pattern shown in figure 4-16. Numbers on the axis indicate decibels below maximum. The directivity index is -22.5 db. This unit consists of 182 nickel tubes spaced in an equilateral triangle. The tubes are premagnetized by permanent magnets.

Directivity patterns of two types of QB transducers are shown in figures 4-17 and 4-18. Figure 4-17 is the pattern of the QBF, an echo-ranging transducer consisting of 450 Y-cut Rochelle-salt crystals mounted on a steel plate. Numbers on the axis indicate decibels below maximum. The directivity index is -25.2 db.

Figure 4-18 shows the pattern of the QBG transducer taken in the horizontal and vertical planes at 22 kc. Numbers on the axis indicate decibels below maximum. The directivity index for the horizontal pattern at 22.5 kc is -17.3 db. The QBG is a small Rochelle-salt gear intended for small ships.

When a transducer is used as a hydrophone, the directivity is generally found to be nearly identical with its pattern when used as a projector, provided the electric connections are equivalent for both sending and receiving. The beam pattern or directivity function *B* (equation (1-13) expressed in decibels) gives information concerning the response of the transducer to sound coming from a specified direction. Even if the sources of sound are uniformly distributed in all directions, the directivity

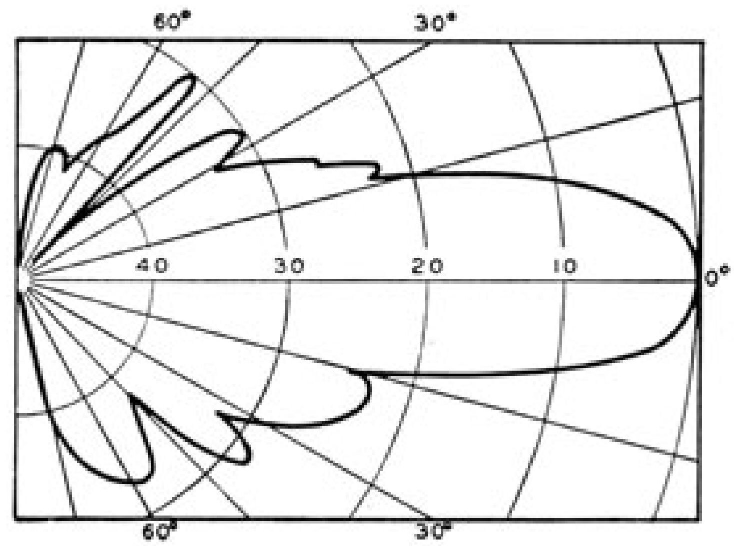
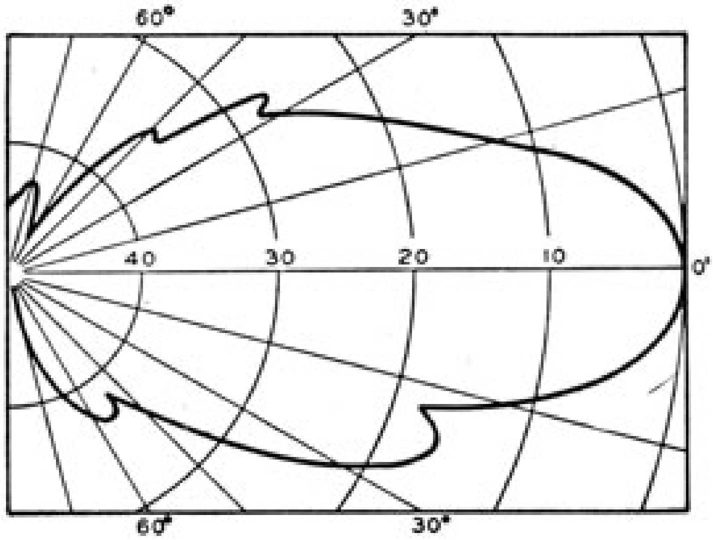


Figure 4-17 -Directivity pattern of Rochelle-salt (Y-cut) crystal echo-ranging transducer (QBF) at 30 kc.

measured by means of a nondirectional hydrophone-that is, one for which b=1 in every direction.

To provide a more accurate bearing determination, the electric connections to the acoustic elements of the transducer may be altered when its function is changed from projector to receiver. One method is to split the transducer elements electrically into two



Directivity pattern of magnetostriction 25-kc echo-ranging transducer (QCU).

function gives information about the transducer to such multidirectional sound fields, because the response is caused largely by those sources in the direction of the main lobe. Sources in other directions do not contribute appreciably.

These multidirectional sounds very often interfere with the reception of echoes. The response of a transducer to these extraneous sound sources, and their previous measurement under various sea conditions and at various locations all become important.

The magnitude of a multidirectional sound field is most readily specified in terms of its rms sound pressure, *p*. This pressure can be directly halves and connect them so that through one amplifier the transducer is most sensitive to sounds coming from slightly to the right of the transducer bearing. Simultaneously through another amplifier, the transducer is most sensitive to sounds coming from slightly to the left. The transducer, as a hydrophone, thus has two directivity patterns, which are not the same as the pattern when the electric connections are not altered.

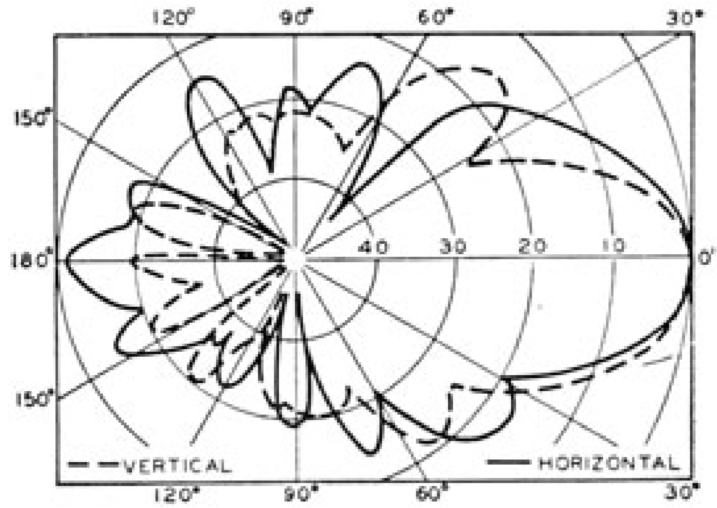


Figure 4-18. -Directivity patterns of Rochelle-salt crystal (45° Z-cut) echo-ranging transducer (QBG) taken in both vertical and horizontal planes.

Directivity Index

The directional characteristics of a transducer can be described by stating the fraction of the sound energy that is sent out in the *desired direction. This fraction is found essentially by computing the directivity* index.

The *directivity factor*, *K*, is the ratio of the total energy radiated by a transducer to the energy that would be radiated if the transducer radiated its maximum intensity in all directions. The directivity factor is also the ratio of the average of intensities taken in all directions to the maximum intensity. This ratio evidently provides quantitative information on the directivity. The directivity factor may be useful also in computing the total acoustic power from an absolute-intensity calibration made upon the principal lobe. If *K* is unity, the transducer is entirely non-directional, whereas if it is a small fraction, a large proportion of the energy is concentrated near the direction of maximum emission, the "acoustic axis."

If the average intensity is bar(I), and the maximum or *axial intensity* is I_a, the directivity index, *D*, is defined by

$D = 10 \log K = 10 \log \bar{I}/I_a$. (4-8)

For a nondirectional transducer, D is zero; for a directional one, D is a negative number. The directivity indices of the various highly directional transducers mentioned in the preceding paragraphs range from -20 to -26 db.

Measurement of the directivity index is required in order to obtain the efficiency of a transducer. It is unfortunate that these measurements are the most difficult and least accurate of all calibration tasks. In the present state of the art very great care is required to obtain an accuracy of ±1 db, and a circular plate with a diameter, *d*, that is greater than 2 wavelengths can be shown to have a directivity factor of

$K = (\lambda/\pi d)^2$, (4-9)

or the directivity index is

$D = 20 \log (\lambda/\pi d)$, (4-10)

where λ is the wavelength in units the same as those of *d*.

Generally, *D* is calculated from the beam pattern, or directivity function, *b*, which was defined by

$b = I/I_a$, (4-11)

where I is the intensity at a given point and I_a is the intensity at a point equally distant from the source but located on the axis. If *b* is averaged over all directions, this average evidently gives *K*, and hence *D*.

When used as a hydrophone, the directivity index of a transducer is defined as follows:

Sound incident on the hydrophone from a standard source located at a point in any direction at a distance *r* from the hydrophone generates electric power W_2. The same source placed on the acoustic axis at the same distance generates electric power W_a. The ratio W_2/W_a can be called b', the directivity function of the receiver. The values of *b* and *b*' are equal for a given transducer; unless, the transducer is split for accurate bearing determination.

As with the projector, *b*' can be averaged over the directivity pattern and the value of *D* calculated as before.

errors of ±2 or ±3 db are much more usual. For this reason and for theoretical reasons it is desirable to obtain an expression for the directivity factor and index.

If simplifying assumptions are made-such as uniform loading, uniform phase and amplitude distributions, and infinite baffle-the directivity index for a transducer of a given size and shape can be calculated theoretically from the constants of the apparatus without involving excessively unwieldy mathematical treatment. For example,

The directivity index of a hydrophone also determines its response to a multidirectional sound source. Consider two sound fields, one caused by a single source located on the axis of the hydrophone, and another by sources distributed equally in all directions from the hydrophone. Let (1) both sets of sources result in the same sound pressure at the hydrophone, (2) E_a be the emf generated by the single source, and (3) E_i be the emf generated by the isotropically distributed sources. Thus

$$20 \log E_i = 20 \log E_a + D. \quad (4\text{-}12)$$

Because D is a negative number, E_i is less than

E_a. This relation has practical importance in the calibration of hydrophones.

IMPEDANCE OF TRANSDUCERS

Mechanical Impedance

Transducers designed for the generation of ultrasonic waves in water have a construction that is markedly different from that of the familiar loudspeakers for the generation of sound in air. It is not possible to do justice here to all the factors entering into these designs, but some of the basic principles are summarized.

The objective, in both constructions, is to set the medium into periodic motion. To set it into motion a force must be applied to the medium. This operation is accomplished most readily by means of a plate, or diaphragm, to which the force is applied almost directly. This plate is often a circular disk. Suppose it is desired to give a point on its surface the velocity

A closer analogy is obtained by considering the motor as a transformer. Then the velocity V_o, is analogous to the output current, and the force F to the output voltage of the transformer. The radiation impedance is directly analogous to the impedance of the output circuit of the transformer. This analogy can be used to describe the effect of taking a transducer out of water and into air. Suppose the transducer has been designed to work under water. Then the lower radiation impedance of air effectively short circuits it. The transducer heats up, just as an ordinary transformer when it is short-circuited. Very little power is usefully transformed.

Conversely, a transducer designed to work efficiently in air is analogous to a transformer with a low-voltage, high-current secondary. Such a transducer is not efficient under water, where the requirements correspond to a high-voltage, low-current secondary.

The physical differences between a loudspeaker designed to work in air and a transducer designed to work in water can be understood by means of this

$v=v_o \cos \omega t$ cm/sec, (4-13)

where v_o is the maximum value of the velocity, ω is $2\pi f$, and f is the frequency of the sound to be produced. If this velocity is attained, the water or air in immediate contact with the plate probably moves with this same velocity when v_o, is not too great. Later in this chapter, the possibility is considered that the medium does not follow the motion of the plate, but for the present such lost motion is ignored.

The first problem is the calculation of the force required to produce the motion. This force is proportional to v_o and to a quantity *Z*. This relation is analogous to that between voltage and current in an electric circuit, and *Z* is, by analogy, called the mechanical impedance of the plate. The resistance and the inductive and capacitive reactance that make up the electric impedance have their mechanical analogies. The value of *Z* depends on (1) the mass, size, and shape of the plate; (2) the mechanical properties of the plate, such as stiffness; (3) the density of the medium; (4) the velocity of sound in the medium; and (5) ω.

The force required to drive the diaphragm or plate at the velocity vo is supplied by an electro-mechanical device called the motor. It is similar to the ordinary motor in that it converts electric power into mechanical motion, but, because the motion is oscillatory rather than rotatory, the analogy is not very close.

analogy. The loudspeaker always has a thin diaphragm of small mass-one that is easily movable. The motor usually applies the necessary force by magnetic means. In principle, a small bit of magnetized steel attached to the diaphragm might be attracted and repelled by a stationary electromagnet through which an alternating current is passed. Even if such a device could be immersed in water without physical damage, the force obtainable in this way would not be sufficient to move the mass of water in contact with the diaphragm, and the device would "stall."

Underwater transducers usually have more massive diaphragms, which are appropriately described as plates. The moving part of the motor is in rigid physical contact with the plate. The large force necessary to move the plate and adjacent water is produced by any one of several methods. It is possible to design electromagnets to furnish this force, but more motors in use at the present time depend on the magnetostrictive of the piezoelectric effect for this purpose. These effects are capable of producing large forces without the complications that would result from the use of large electromagnets.

SOUND OUTPUT

Electric Power Input and Acoustic Power Output

In rating a transducer, it is essential to know how much of the applied electric power is available as acoustic power, and how much of the available acoustic power is concentrated in a narrow beam.

The electric power input can be measured either from the applied voltage and the impedance of the transducer or from the current and impedance of the transducer.

The acoustic power output can be computed from measured pressure levels. The total power is given by the energy flow per second through the surface of a sphere surrounding the transducer. The average intensity bar (I), over a sphere of radius r multiplied by the surface area of the sphere, $4\pi r^2$, therefore is a measure of the acoustic output of the transducer. Because bar(I)=KI_a, where K is the directivity factor and I_a is the axial intensity, the acoustic power is $4\pi r^2 KI_a$.

The axial intensity is commonly measured by mounting a hydrophone at a convenient distance on the acoustic axis of the transducer and transmitting continuous sound by use of a constant signal current.

Efficiency and Response of a Transducer

The performance of a given transducer is completely described by the response, the directivity index, and the efficiency. The characteristics of some standard echo-ranging transducers are listed in table 7.

TABLE 7 - *Characteristics of Some Standard Transducers*

Code	Type	Resonant freq. (kc)	D	Source level (S_a)	Efficiency (db)
QGA-942	MS[1]	30	-23.2db	85db	-6
QGA-941	MS	15	-18.1	77	-7.5
QBF	RS Y-cut	24	-21.1 (20kc) -23.5 (26kc) -25.2 (30kc)	88.5	-3.6
QBG	RS X-cut	22	-17.3 (22 kc)	33 (22kc) 39 (45kc)	?
QCU	MS	25	-22.5	84	-3.8
QCL	MS	20	-21.4	43	-9.5
QCJ	MS	24	-22.1	46.5	-9.5
Asdic	Quartz	15	-22.0	56	-3.1

[1] Magnetostriction.

Limitation of Power Output by Electric Characteristics

It would appear that very long echo ranges might be achieved by increasing the power input into the transducer system, and that the only limit on the available power would be imposed by the permissible size and weight of the gear. This supposition is not true. There are two limiting factors in determining the power output, aside from structural requirements.

The first of these factors results from electrical characteristics. The voltage across the face of a crystal cannot be increased

Only that portion of the electric power that is converted into acoustic power is available for echo ranging. The efficiency of a transducer is defined in decibels by 10 log (P_o/P_i), where P_a is the acoustic power output and P_i is the electric power input. If a system is, for example, 50 percent efficient, the efficiency is 10 log ½, or -3 db. An efficiency of 10 percent would be -10 db, and so on. The efficiency of a standard echo-ranging transducer ranges from -2 db to -15 db.

A convenient method of rating a transducer is to state the axial sound level reduced to 1 yard[1] (the axial source level) per volt or ampere of the impressed voltage or current. This value is called the response of the transducer.

The acoustic power output *P*, the axial source level S_a, and the directivity index *D*, are related by the empirical equation

$$S_a = 71.6 + 10 \log P - D. \quad (4\text{-}14)$$

indefinitely, for at a certain critical voltage a spark passes. This action is referred to as *voltage break-down*. Some idea of the magnitude of the maximum voltage that can be applied may be gained from the fact that the specifications for ADP crystals for echo-ranging transducers require that each crystal must withstand a voltage gradient of 20,000 volts per inch at a frequency approximately one-half the resonant frequency for at least 30 seconds.

In magnetostriction transducers a limitation to the power input is set by the fact that the magnetostriction effect becomes negligible when a certain critical value of the magnetic field strength

---

[1] The standard unit distance for calibration adopted by the Navy is 1 meter. One yard and 1 meter are not sensibly different for this purpose.
239276°-53-7

is reached. Nickel, for example, exhibits practically no magnetostriction for field strengths greater than from 200 to 250 gausses. (See figure 4-2.)

Limitation of Power Output by Cavitation

The second factor that limits the power output of transducers is cavitation.

An acoustic transducer consists essentially of a vibrating face or piston. The motion of the face is imparted to the water, in which the disturbance is propagated as a wave. This process can proceed efficiently only as long as the water follows the motion of the transducer face. When this motion becomes too violent, the face tears away from the water, with a marked loss of efficiency in the process of sound production.

This limitation on the output of a transducer is thus closely related to the phenomenon of cavitation. Let p be the rms acoustic pressure at a point where the normal hydrostatic pressure is p_o.

Then once each cycle of the sound wave the total pressure changes from p_o-1.41p to p_o+1.41p and back again. Cavitation may occur whenever the total pressure tends to become negative. Accordingly, the greatest rms acoustic pressure that can be transmitted through a region where the hydrostatic pressure is p_o is $p=p_o/1.41$. In terms of sound level,

critical level= 20 log p_o-3. (4-15)

warmer than the water. This formation may occur also under other conditions. Accompanying the formation of these almost invisible bubbles, the sound output of the transducer for a given electric input is much reduced. Under similar circumstances its sensitivity as a hydrophone also diminishes.

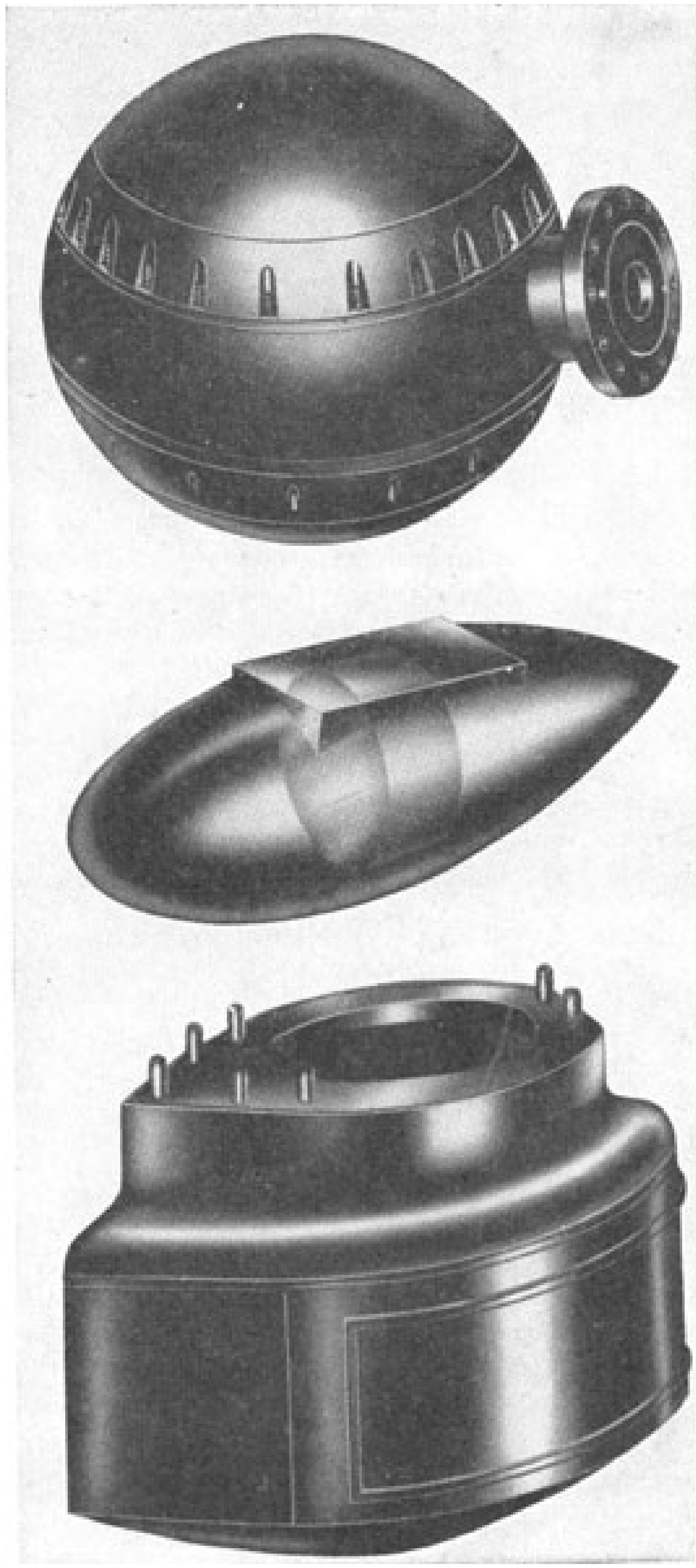


Figure 4-19 -Standard domes.

When p_o is 1 atmosphere (35 feet of water or 10^6 dynes/cm^2), L is 117 db. When the sound level exceeds this critical value, cavitation bubbles may be formed and cause high transmission losses. Cavitation bubbles are described with wakes in chapter 2.

Because the acoustic pressure is highest at the face of the transducer cavitation occurs there before it occurs elsewhere. This action constitutes the process discussed earlier. As a result of the process, the power output of the transducer, for a given motion of the face, is reduced.

Aside from the reduction of power output of a transducer for a given motion of its face because the water does not follow the moving face, the power output may be reduced for other reasons. Thus, it has been observed in experimental tanks at the Naval Research Laboratory that small air bubbles may form on the transducer when it is

EFFECT OF DOMES

The transducer unit, consisting of the transducer and the shaft that supports it, is usually installed near the bow of the ship. Because the housings in which transducers are encased are usually spherical, they cause excessive turbulence, and sometimes cavitation, even at moderate speeds. This action causes excessive background noise. For this reason, transducers are generally enclosed in streamlined metal shells

place, the directivity pattern of the dome-enclosed transducer differs from that of the same transducer without a dome.

The two effects are closely related. It is not possible to construct domes of materials that are entirely transparent acoustically. Thus, a certain amount of multiple reflection occurs inside the dome. As a result, some of the sound energy that is emitted by the transducer into the main lobe of the sound beam is diverted from it. This action reduces the axial source level.

Any energy diverted from the main lobe, however, must be redistributed in some manner. It is quite possible, therefore,

called domes.

Several types of domes are in current use by the Navy. They are all made of corrosion-resistant steel; the front is very thin so as to form a "window" to transmit the sound; the back is heavy so as to damp unwanted noise from the propellers. One type is equipped with a bulkhead just aft of the transducer, which supports a sound-absorbing baffle on the forward side and a sound-reflecting pad on the after side. Both these devices reduce sound reception through the stern section of the dome, and the baffle also aids in reducing multiple reflection within the dome. Some domes are retractable and when not in use are withdrawn into a sea chest built into the hull.

Some standard domes are illustrated in figure 4-19.

The acoustic effects of the use of domes are two-fold. In the first place, the axial source level of a dome-enclosed transducer is less than that of the same transducer without a dome. In the second that new side lobes may be added to the directivity pattern, for the regular shape of the dome would preclude a mere random redistribution of the diverted energy. Moreover, it is obvious that multiple reflections inside the dome may affect the original side lobes of the bare transducer pattern.

The decrease in the axial source level due to the distortion of the directivity pattern is equal to the change in the directivity index that ensues when the transducer is placed in a dome.

In echo ranging, a loss in the transmission reduces the effective range; and the distortion of the directivity, especially if accompanied by the formation of prominent side lobes, tends to confuse the determination of bearings. Hence, the various factors that have been adduced must be considered in the design of a dome.

Receiver Sensitivity and Background Noise

RECEPTION

Although the several methods of echo portrayal are quite different, the general principles that govern them are similar. The echo is only one of many sounds picked up by the sonar. Each sound, whether wanted or unwanted, actuates the portrayal device. The echo must be heard in spite of the unwanted sounds that are being heard at the same time, or it must be seen among the records of these other sounds.

An ideal sonar would respond only to the echo and not to any other sound. This ideal is unattainable, but steps can be taken to approach it. For example, in listening to the radio we wish to hear the broadcast of only one station at a time; so we tune our set, with the result that it responds only to the electric waves of the relatively narrow range of frequencies emitted by the particular station and not to those of any other. In the same way, because the echo has a definite frequency, it obviously is desirable to tune the receiver to this frequency, thus excluding much of the unwanted sound. Such tuning is more important with visual than with aural methods of portrayal, for the ear has the ability to ignore unwanted sounds and to hear a note of definite pitch even in the presence of noise.

The sonar receiver can be tuned at various stages. The first is the so-called radio-frequency (r-f) stage, in Which the receiver can be tuned to the frequency of the incoming echo. In the

second stage, the receiver is tuned to the intermediate frequency (i-f), which is the first heterodyne stage. Finally it is possible also to tune the receiver in the audio-frequency (a-f) stage, where the once-heterodyned signal is heterodyned a second time to an audible frequency. The tuning is under the control of the operator, and can be accomplished at any one stage or in several stages at once.

Another approach is found in the fact that the echo is sound coming from a particular direction, whereas background noises may come from all possible directions. The unwanted noise can be reduced by using directional transducers. The obvious disadvantage of such a receiver is that it cannot then be alert in all directions simultaneously; but this

Response Curves

The graph showing the response at each frequency is called the response curve of the system. The *response curves* of two QC magnetostriction transducers are shown in figure 4-20. They

drawback is offset by the consideration, that, if an echo is received on a directional sonar, the bearing of its source is known at once.

RESPONSE OF TRANSDUCERS AND AMPLIFIERS

Response of Transducers

The electromotive force generated by the transducer is a function of the sound pressure on its diaphragm. This response of the transducer partially determines the response of any system into which it may be connected. Transducer sensitivity at the frequency, *F*, is defined as the emf developed in the transducer when it is in a sound field of frequency, *f*, and in a rms pressure of 1 dyne/cm^2. If *e* is the emf generated by the transducer when in a sound field of *p* dynes/cm^2, the ratio

$k=e/p$ (4-16)

defines the sensitivity of the transducer. It is measured in volts/dyne/cm^2. The quantity

$K=10 \log k^2= 10 \log e^2/p^2=20 \log k$, (4-17)

is called the response of the transducer. The response is the decibel ratio of the power generated by the transducer per ohm resistance of the external circuit to the intensity of the sound field at the transducer. If *P* is power per ohm resistance and *I* is the intensity of the sound, then because $P=e^2$ and, *I* is proportional to p^2

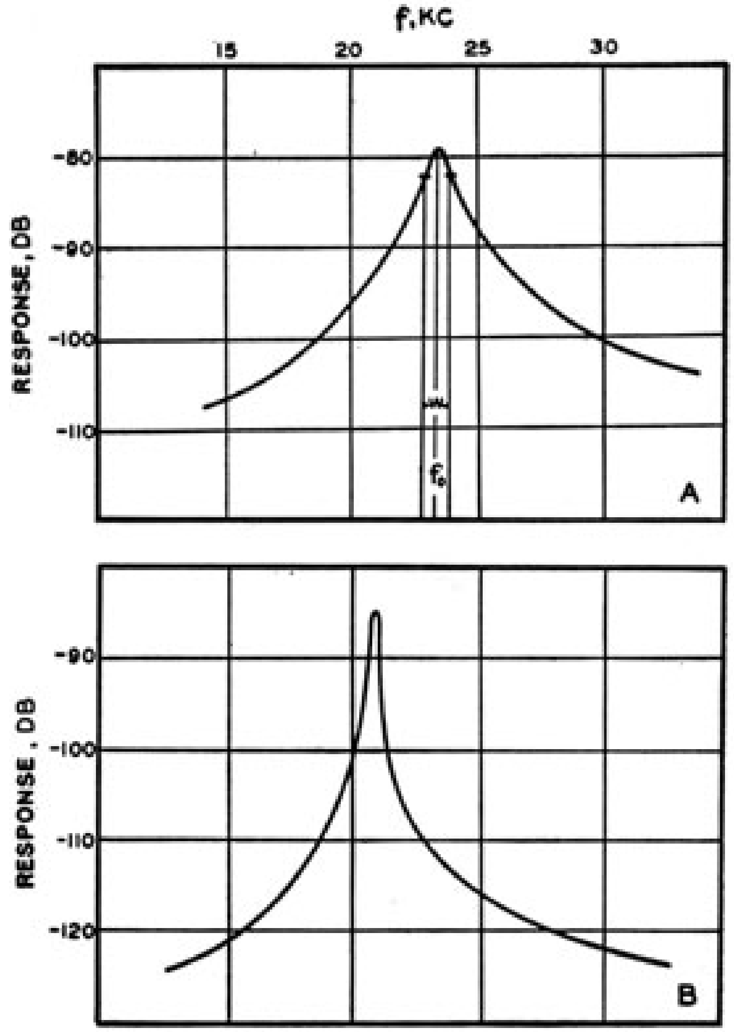


Figure 4-20 -Response curves of transducers. A, Type QCJ; B, type QCL.

respond well only to sounds in the neighborhood of the resonance frequency, F_o. In the case of the QCJ (figure 4-20, A), F_o is 24 kc-that is, the transducer is said to *resonate* at 24 kc. The width, *w*, of the resonance peak, shown in the figure, is usually defined as the frequency separation of the two points on the curve which are 3 db below the maximum. In the given curve, *w* is about 1 kc.

Another commonly specified quantity is the resonance parameter, $Q=F_o/w$. If *Q* is greater than about 20, the system is said to be highly resonant; if *Q* is less than 4 or 5, the system is nonresonant. The QCJ transducer has a *Q* of about 24. The QCL shown in figure 4-20, B, is more sharply.

K=10 log e^2/p^2=10 log P/I. (4-18)

resonant than the QCJ; the *w* of the QCL is between about 200 and 300 cycles, and, because it resonates at 21 kc, its *Q* is between 70 and 105. Because of the resonance of these transducers they are tuned-that is, the echo frequency must be near the resonant frequency or they will not respond effectively.

Response of Amplifiers

The amplification ratio of an amplifier is similar to the transducer sensitivity. It is the ratio of the output voltage to the input voltage. The response is defined in terms of amplification ratio in exactly the same manner that the response of a transducer is defined in terms of its sensitivity. Response curves can be plotted for amplifiers as well as for transducers and the same terminology is applied to them.

SPECTRUM LEVEL AND RESPONSE TIME

Power Spectrum Level of Noise

The response curve shows the emf generated by a transducer in responding to a sound of a definite frequency. Most of the unwanted sounds encountered in echo ranging do not have a definite frequency, and it is necessary to consider the emf generated by the transducer in response to such a sound.

Consider an ideal transducer the

call that of a continuous noise a power spectrum and that of a pulse, an energy spectrum. Equation (4-19) then becomes

10 log P=K+N+ 10 log w. (4-21)

For all wide-band noises encountered in echo ranging, equation (4-19) is sufficiently accurate even for resonant transducers like the QC, the response curves of which are far from ideal. Equation (4-19) indicates that the power generated by a wide-band noise is proportional to the width of the resonance peak of the transducer.

Energy Spectrum Level of a Pulse

Although the intensity of an uninterrupted, constant sound is most conveniently measured in terms of power (energy per second), the intensity of a pulse is better measured in terms of energy- that is, power times duration. The energy spectrum of a pulse can be defined in much the same manner as the power spectrum of an uninterrupted sound.

If the pulse consists of a train of sinusoidal waves, it will have a fairly definite pitch, say F cycles per second provided the train contains many complete waves. The definite pitch of such a pulse indicates that its energy spectrum has a sharp maximum at the frequency F. If the number of waves in the train is diminished, the height of this peak decreases, and its width w increases. The complete mathematical discussion of this effect can be given in elaborate form, but the essential result is simple.

Let the duration of the wave train be τ seconds. Each wave requires $1/F$ second to pass a given point. Therefore, if the wave train contains many complete waves, $\tau \gg 1/F$ The duration of the pulse

response curve of which is rectangular, as illustrated in figure 4-21. Suppose that it is possible in some way to vary both F_o and w, while the transducer is exposed to a constant noise. The emf generated then depends on both F_o and w. If w is made successively smaller, the power P of the generated emf finally becomes proportional to w-

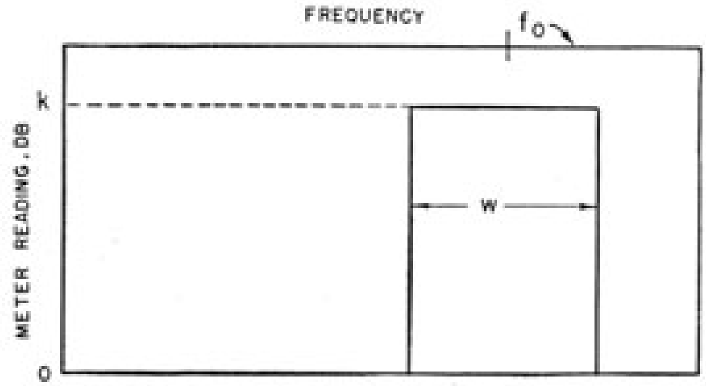


Figure 4-21 -Response curve of an ideal transducer.

$P=k^2 I(F_o)w$. (4-19)

The two other factors in this equation are k, the sensitivity of the transducer to a sound of the frequency F_o, as defined by equation (4-16) and a function $I(F_o)$, which is characteristic of a particular noise. This function has not been given a simple name but is sometimes called the intensity of the noise in a 1-cycle band. The function

$N(F_o)=10 \log I(F_o)$ (4-20)

is called the spectrum of the noise, or its *spectrum* level at F_o.

To distinguish the spectrum of a continuous noise from that of a pulse, it is often necessary to

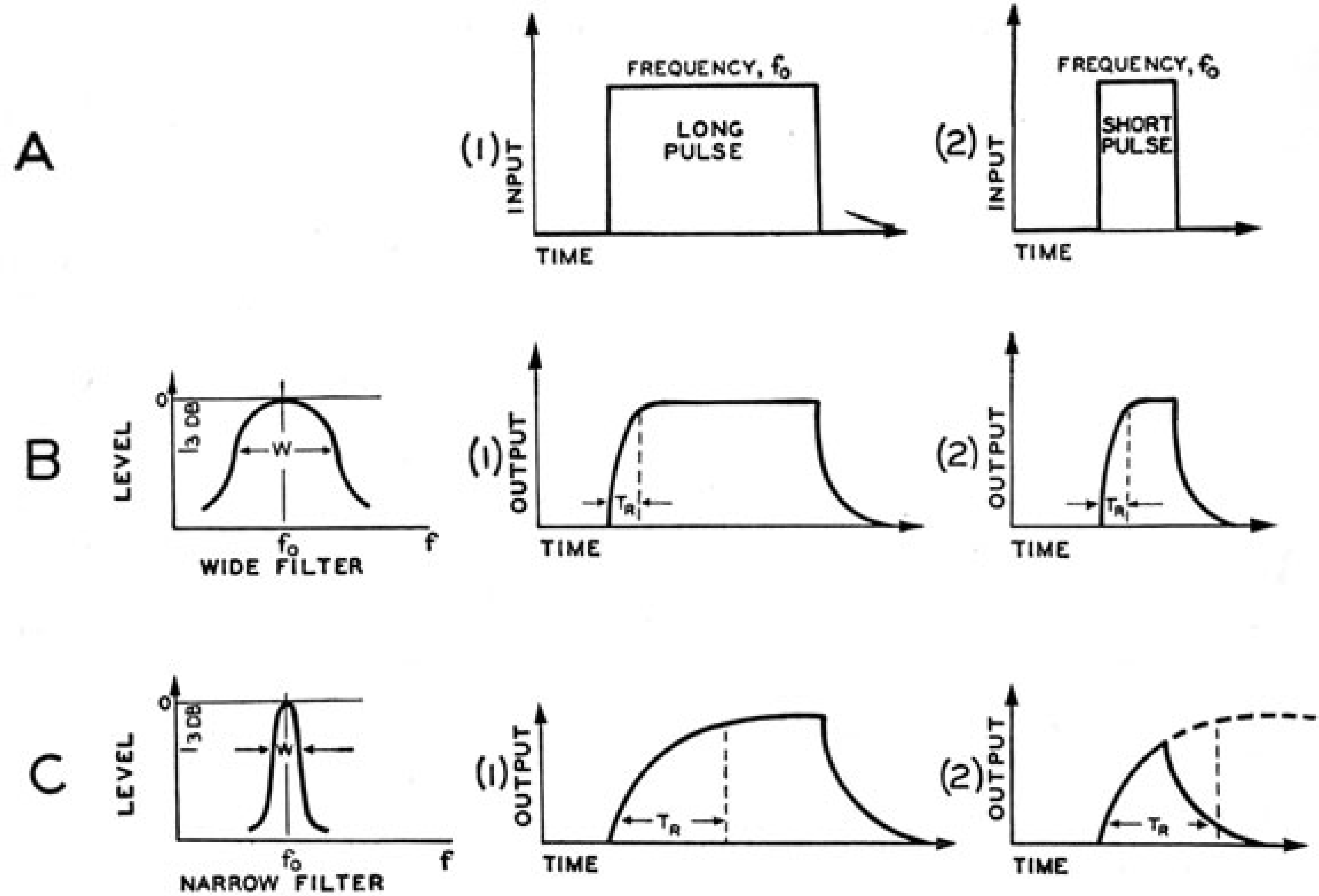


Figure 4-22 -Response time of a filter in relation to its bandwidth.

τ and the width w of the resonance peak of the pulse are connected by the approximate equation

wr=1. (4-22)

The greater width of the resonance peak associated with a shorter pulse duration makes it appear that a short pulse can be analyzed into a much wider group of frequencies than a long one. The human ear behaves in a manner consistent with this mathematical relation. If a listener hears pulses consisting of trains of sinusoidal waves, his sensations depend on the number of waves in the train. If the pulse contains many complete waves, the sensation is that of a short tone of well-defined pitch. As the number of complete

Response of Band Filters to Short Pings

Because the width of the spectrum peak of a pulse is inversely proportional to the pulse duration, it might be expected that in designing filters intended to pass only a restricted group of frequencies centered at *F* cycles per second the duration of the pulse would have to be considered. As a very short pulse has a wider peak it seems obvious that if a filter is to pass it with minimum diminution of intensity the width of the filter must be greater than is necessary for a longer pulse with its proportionally narrower peak.

This fact can be stated in another way. There is a relation between the width of the filter and the speed with which it responds to sudden changes of input. This relation is illustrated in figure 4-22. The two upper graphs, *A* (*1*) and *A* (*2*), represent the input of a long and a short pulse, respectively. The graph at the left in row *B* shows the

waves diminishes and the pulse becomes shorter the listener finds it more and more difficult to be sure of the pitch. Finally, very short pulses consisting of only two or three waves lose all tonal characteristics and are best described as "clicks" or "pops."

frequency response curve of a wide

filter; the one at the left in row *C*, that of a narrow filter. The remaining curves-*B* (*1*), *B* (*2*), *C* (*1*), and *C* (*2*)-represent the outputs of the filters when excited by the corresponding pulse shown in *A* (*1*) and *A* (*2*).

The input in each case begins and ends gradually. It requires a certain time interval, t_R, to come anywhere near its maximum value. theoretically, it requires an infinite time to reach its maximum value; hence the rather vague working of this sentence. This time interval is indicated in each of the diagrams, which show that it is much shorter in the wide filter, *B*, than in the narrow one, *C*. The relation between the response time, t_R seconds, and the width, *w* cycles per second, of the filter is given by the inequality

t_R greater than or equal to 1/*w*. (4-23)

In a well-designed filter the equality may be assumed to hold; but a poorly designed filter may have a response time considerably greater than 1/*w*.

The diagrams show that if the pulse is long enough, as illustrated by the curves *B* (*1*) and *C* (*1*), the response time is short

eliminating the unwanted sounds. Several of these factors will be discussed briefly.

Doppler Effect

Even if the frequency of the emitted signal remains constant, the frequency of the echo is not always the same, but depends on the rate at which the range of the target is changing. Therefore, the tuning cannot be made indefinitely sharp without endangering the reception of echoes from targets with a high range rate.

The change in frequency due to the Doppler effect can be very great.

In chapter 3 it was shown that if the emitted frequency in kilocycles is F_o, and the range rate in knots is *dR*, the frequency of the echo is changed by *f* cycles per second. Because the Doppler effect is to shift the frequency by 0.7 cycle per kilocycle per knot of range rate,

$f = 0.7\, F_o dR$ cycles per second. (4-24)

This change, *f*, is an increase if the range is closing and a decrease if it is opening. Because the sonar vessel has a usable speed of 20 knots and the possible targets have speeds of 20 knots, the range rate can be 40 knots opening or closing. If the sonar vessel is emitting 24-kc sound, the doppler shift is 672 cycles per second. This condition requires that the band pass be twice that width, or 1,344 cycles per second. To overcome this difficulty (as previously mentioned), a device called *own doppler nullifier* (ODN)

enough that the pulse can come near its maximum value even in the narrow filter, *C* (*1*). If the pulse is short, however, as illustrated by the curves *B* (*2*) and *C* (*2*), the response time of the wide filter is short enough to permit the pulse to come up to maximum value *B* (*2*), but this principle does not apply to the narrow filter, *C* (*2*). In *C* (*2*) the input ceases before the response time has elapsed. Thus the output is always less than its maximum value.

It follows that if a receiver is to respond fully to pings of a length r_o yards, the duration of which is thus

$2r_o/v = r_o/800$ seconds,

where v is 1,600 yd/sec, the pass band of the receiver must be at least $800/r_o$ cycles per second.

LIMITATIONS ON THE USE OF SHARPLY TUNED RECEIVERS

Besides the limitation on filter width set by the ping length, there are several other factors that prevent full exploitation of tuning as a method of

and *target doppler nullifier* (TDN) is used.

This device automatically adjusts the output of the receiver to the desired frequency, for example, 800 cycles per second. These doppler nullifier circuits are shown in chapter 7.

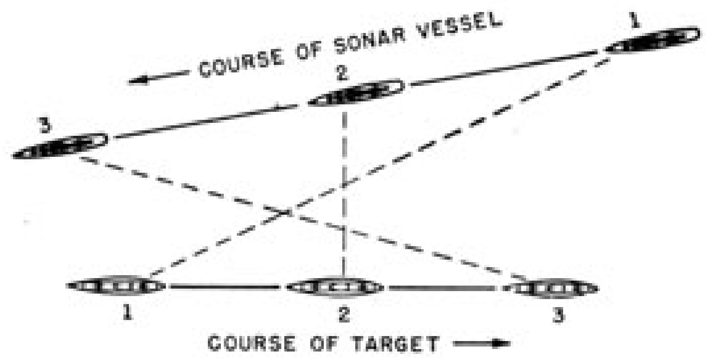


Figure 4-23 -Relation of the range rate to the relative bearing of the target.

The range rate depends not only on the speeds of the sonar vessel and the target, but also on the relative bearing of the target. This principle is illustrated in figure 4-23, which shows three successive positions of two vessels passing on constant courses at constant speeds. At time 1 the target is off the port bow of the sonar vessel, and the range is closing rapidly. At time 2 the target is at the point of closest approach and the range rate is zero. At time 3 the target is on the port quarter and the range is opening rapidly. Thus in this case, the change from closing to opening range occurs continuously as the bearing changes.

Reverberation

Reverberation occupies a peculiar position, as it is in some ways an unwanted sound and in others a wanted sound. From the standpoint that reverberation can mask the echo, it is unwanted. Considered in this light, it unfortunately has a frequency that is very close to that of the echo, so that extremely sharp tuning is needed if the receiver is to respond to the echo and not to the reverberation.

Reverberation consists essentially of a large number of echoes; hence its frequency also is affected by doppler. Because the scatterers responsible for the echoes are presumably at rest in the water, the range rate at which the vessel is approaching these scatterers and the magnitude of the accompanying Doppler effect are determined solely by the speed of sonar vessel and the relative bearing of the sound beam. For this reason the Doppler effect of reverberation is called by ear, the masking power of reverberation is reduced. Reverberation then functions primarily as a wanted sound. On the other hand, when the speed of the target through the water is small or at right angles to the transducer heading, the unwanted masking effect of reverberation is dominant. This unwanted effect is always the dominant one with visual methods of portrayal, because the range recorder cannot distinguish between sounds of various frequencies.

Own Doppler Nullifier

The major limitation on the use of narrow-band receivers is the necessity for allowing echoes of many different frequencies to pass through the receiver. The limitation would not be so severe if the sonar vessel were at rest, or if its motion did not affect the frequency of the echo. The speed of most targets is relatively small, and the width of the receiver band could be reduced if target speed were the only cause of Doppler shifts.

It would be possible to operate with a narrow-band receiver if the operator could constantly change the frequency of the transmitted signal by an amount just sufficient to compensate for own doppler. The reverberation frequency would then remain constant, and the only frequency shift to be accommodated would be that of target doppler. Because of the necessity of sweeping the sound beam over a wide range of bearings, the own doppler changes rapidly, and it is not feasible for the operator to make this adjustment and still perform his other duties. The ODN automatically accomplishes this result to a high degree of approximation.

When the ODN is set for some one frequency, for example, 800 cps, the local oscillator frequency is automatically adjusted to give this value during the first fraction of a second after transmission; thereafter the adjustment remains constant. In this way the frequency of the reverberation is kept quite constant, and considerably narrower filters can be used in the receiver. The ODN is less useful against submarines capable of high underwater speeds. The TDN is employed to return any echo to 800 cycles at the receiver output.

own doppler. If an echo is received from a moving target, there is a difference between the echo frequency and that of the reverberation. This difference is called *target doppler*. The magnitude of the target doppler is an important factor because it is a measure of the speed and approximate relative bearing of the target. From the standpoint that it enables the determination of target doppler, reverberation is a wanted sound.

When the difference in frequency between echo and reverberation is large and the echo is detected

Sonar Location

CONSIDERATION OF VESSEL HULL DESIGN

Sonar equipment is installed in many types of vessels ranging from small patrol craft to large

aircraft carriers and submarines. Each type of hull presents different problems of sonar installation, depending on its width, length, draft, material, and

---

construction. Although wooden hulls have a greater attenuation for hull-born noise, they have several disadvantages. They are usually very short, are of shallow draft, and are not a stable platform for sonar equipment. The best sonar platform is a long, deep-draft hull.

Selection of Transducer Location

There are no fast rules for the location of a sonar transducer. However, certain factors must be taken into consideration. Because the screws are the primary source of noise, the transducer should be located as far from them as possible. In most deep-draft

There are two reasons why the transducer is not located at the bow. First, the bow does not provide enough width for mounting a standard sonar dome, and second it is not stable. On small craft it does not require a heavy sea to cause the vessel to pound so that the keel section near the bow is "twixt wind and water." Even if this condition didn't cause mechanical damage to the transducer it would cause too many of the transmissions to be quenched.

Use of Fairing on Hull To Reduce Self-Noise

Any protuberance ahead of the transducer probably is a source of noise; therefore, such things as the pitometer log and water intakes should be aft of the transducer. Loose or badly formed rivets are a source of noise. From these statements it should be clear that the flow of water along the

vessels, the screws are high above the keel, so that if the transducer is located well forward, the hull acts as a shield or baffle and allow, the equipment to look in all directions with less interference from the screws.

Another reason for locating the transducer well forward and deep is because of the pressure gradients created by the bow wave.

The pressure of the water around the bow is increased and as the water moves aft the pressure decreases toward normal. When the pressure of sea water is reduced it begins to gas, causing the water to become bubbly. If the transducer were located in or aft of the point at which this condition occurs, a great number of bubbles would strike the face of the transducer with about the same effect as small steel pellets and thus create a very objectionable water noise.

The magnitude of the bow wave depends on the speed of the vessel. Therefore, its normal tactical employment must be considered when selecting the transducer location. Because the normal tactical speed of the vessel is known the normal magnitude of the bow wave is known, and the Neidemair formula can be used to locate the transducer in the high-pressure area. According to the Neidemair formula, the number of feet aft of the bow that the transducer should be located is determined by KV^2, where V is the ship's speed in knots and K is a constant with an average value of 0.14.

hull must be as smooth as possible and any possible point of turbulence must be smoothed.

Fairing is used on hull openings below the waterline to smooth the flow of water. It is made of an elevated,. half-round section of metal that surrounds the opening. It may be either circular or shaped like a tear drop.

Reduction of Noise from Local Machinery

All machinery aboard ship contributes to the over-all self-noise of the vessel. Because the machinery is mounted on the hull and because the hull is a good conductor of sound, the transducer must be insulated from the hull in order to reduce direct coupling of this noise into the sonar equipment. The transducer is generally insulated by a rubber gasket. A noise survey of the machinery may disclose that certain pumps, motors, or generators are excessively noisy. If after inspection the noise cannot be laid to mechanical trouble in the noisy equipment, it should be determined whether the machinery can be mounted so that the noise it generates is not directly coupled to the hull.

The problem of local noise differs from ship to ship and requires individual noise surveys.

CHAPTER 8

SONAR TRANSMITTERS

Introduction

The transmitters used in sonar are conventional amplifiers operating in the low radio-frequency range. In sonars of recent design, the signal to be transmitted is generated in the unicontrol oscillator system at the desired frequency and then delivered to the input of the intermediate power amplifier of the transmitter. In the transmitter this signal is amplified to the desired power level and then delivered to the transducer.

In some of the older searchlight equipments that are still in use the unicontrol oscillator system is not used. In these equipments a master oscillator associated with the transmitter develops the signal at the frequency used in transmission. An example of this system is the model QGB.

In scanning systems a peak-power method of transmission is used to deliver the necessary high energy required to ensure that the power radiated in all directions is equal to the power radiated by a searchlight type of equipment along its selected bearing.

QGB Transmitter Power Amplifier

The circuit of the QGB transmission system comprises a master oscillator and a push-pull power amplifier. This combination (figure 8-1) is capable of delivering 400 watts of power at the frequency determined by the oscillator, which ranges from 17 to 27 kc.

Tuning is accomplished by the tank coil variometer, L101. This inductance, in conjunction with capacitors C101, C102, C103, and C122, determines the output frequency. Under some operating conditions it is desirable to "sweep" the oscillator frequency to provide a more easily distinguishable note.

A motor-driven variable capacitor, C115, is substituted for C122 in order to accomplish this sweeping. This variable capacitor is connected in the tuning circuit by switch 5107, which energizes relay K109. Under normal operating conditions switch S802, located on the control console, also accomplishes this purpose.

The transformer, T102, is of the closed-core type, which with adequate external shielding, minimizes the induction field that might otherwise cause interference with other services. The use of pentode amplifiers results in an economical tube complement because of low-drive requirements. The resistance-choke combinations, Z101 and Z102, in the plate leads of the output tubes, serve as parasitic suppressors. The output-transformer secondary circuit is fixed-tuned and couples energy from the power amplifier to the projector, through contacts of keying relay K105 and the filter junction box. The fixed capacitors of the filter junction box tune out the inductive reactance of the projector and the leakage reactance of the output transformer. The only adjustment required in the output circuit is the selection of the proper capacitors in the filter junction box for the particular projector selected.

The master oscillator tube is biased below the cut-off point, and normally the power-amplifier tubes are likewise biased. Keying is accomplished by

The master oscillator, V101, is capacity-coupled to the power amplifier. The power-amplifier stage, V102 and V103, is coupled to the projector by means of a special high-frequency transformer, T102, which operates efficiently over the tuning range of the oscillator.

means of relay K105. One of its contacts short-circuits the master oscillator bias supply, thus allowing the tube to function. The output of the master oscillator is sufficient to overcome the

2392700-53-12

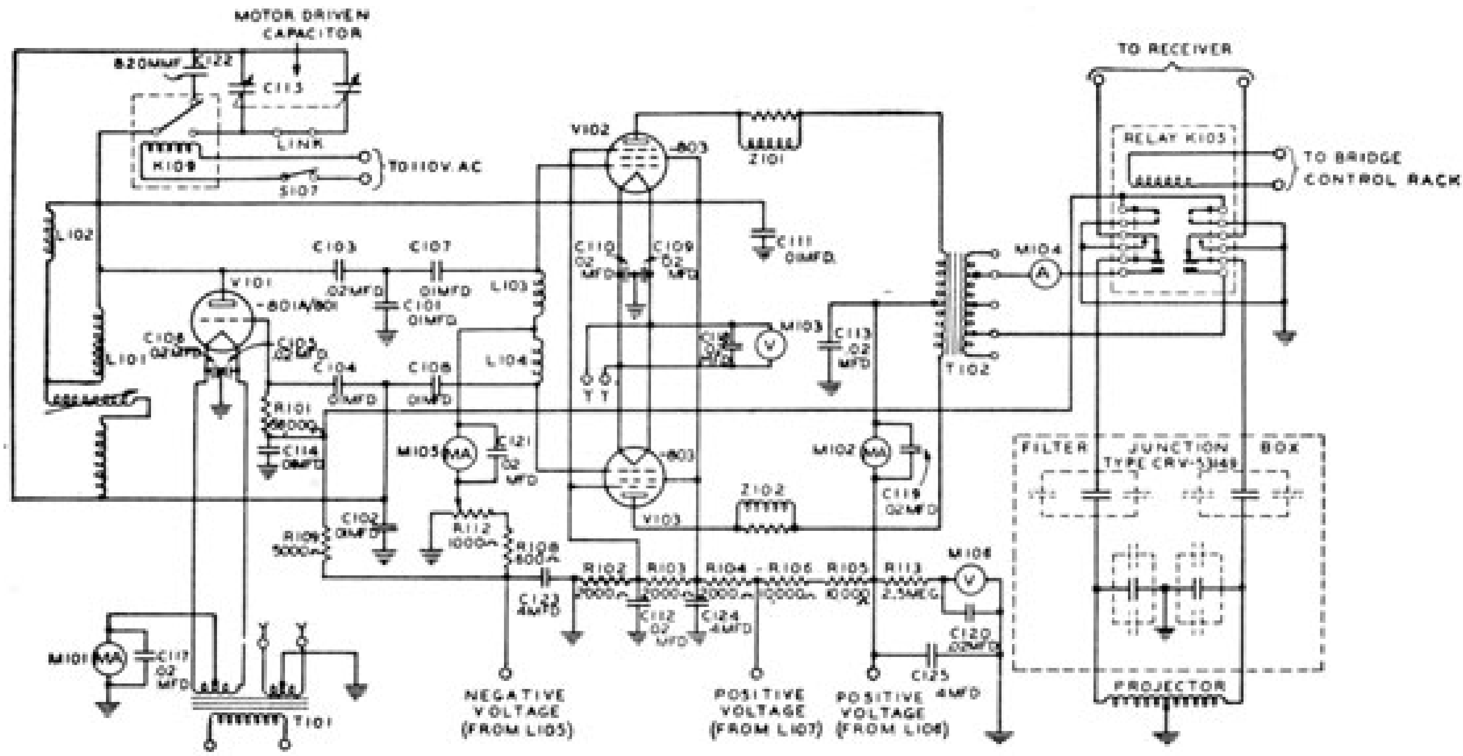


Figure 8-1. -Simplified diagram of transmitter and output circuit.

cut-off bias of the class-C power-amplifier tubes, allowing them to conduct. Keying relay K105 is controlled from the keying unit located in the bridge control rack.

In addition to removing the blocking bias from the master oscillator, the keying relay connects the projector (filter junction box) to the transmitter during its period of operation. When the driver is not delivering power the keying relay connects the projector leads to the receiver. During transmission of the signal impulse, the receiver is disconnected from the projector, and

contacts keep the motor energized until the capacitor has rotated *one complete revolution*, so that the plates are fully meshed, or are at maximum capacity, at the time of starting. Positioning the plates in this manner permits the capacitor to rotate from maximum-capacity to minimum-capacity position coincident with the start and finish of each oscillator pulse.

This variable capacitor causes the oscillator frequency to sweep from approximately 400 cycles per second below to 1,600 cycles per second above the unswept frequency of the oscillator-that is, the

its input is grounded to protect it from the high energy in the projector circuit.

The sweep-frequency modulator (figure 8-2) contains (1) a variable capacitor, C115, which is rotated by an "Alnico" type motor, B101; (2) a fixed capacitor, C122; and (3) a motor capacitor, C126, and resistor, R115, which are associated with the operation of the motor. The motor starts quickly, and when power is removed, the rotor generates a current in the windings. This self-generated current provides electric braking.

A cam-operated switch is mounted on the end plate of the variable capacitor. The cam is attached to the capacitor shaft and rotates as the rotor plates of the capacitor are rotated. This cam actuates two separate pairs of contacts, which complete the a-c circuit of the motor. These frequency at which it operates with the sweep-frequency modulator off.

On becoming energized, sweep relay K110, closes contacts *C* and *D*. This action completes the a-c circuit to the motor through contacts F of the cam-operated switch. The completion of this circuit causes the motor to operate and rotate the variable capacitor and the operating cam of switches *F* and *G*. Because of its shape, the cam causes limit switch *G* to close its contacts after the cam has begun rotating. After a capacitor rotation of 180°-or to the unmeshed position of the capacitor-the cam causes limit switch *F* to open and the motor stops if relay K110 has not been de-energized. Relay K110 is energized by d-c pulses received through the keying relay. These pulses have a definite duration followed by an

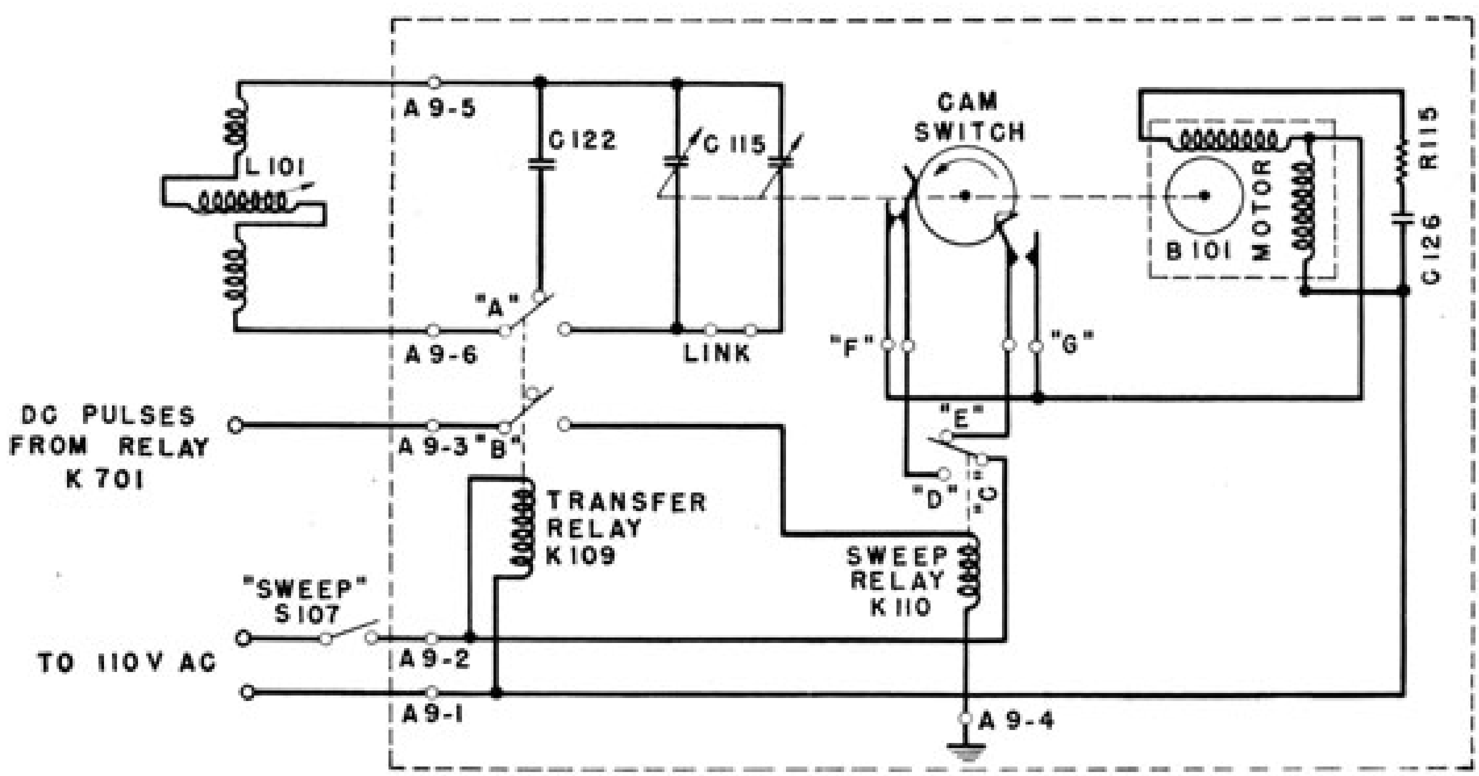


Figure 8-2. -Sweep-frequency modulator.

"off" interval. During the "off" interval, the relay is de-energized, causing contacts *CD* to open and *CE* to close. The a-c circuit to the motor is completed again, this time through the closed contact *G* of the cam switch and the *CE* contact of sweep relay K110. The motor rotates the capacitor and cam another 180° to the starting point of maximum capacity at which time contact *G* of the cam switch opens and the motor stops. cam switch *F* is closed and ready for the next keying impulse that energizes K110, causing the motor circuit to close through contacts *CD* and *F*. Under normal keying conditions the "sweep" is energized momentarily and the sweep capacitor rotates 360° before stopping. The cycle then repeats the next time the equipment is keyed.

QHB Transmitter

The receiver converter of the QHB is used for transmitting and receiving. During transmission it provides both the circuit for producing the signal frequency to be transmitted and the circuits to provide the keying. The 65-kc master oscillator, one-half of V721 (figure 8-3), is the basis of the unicontrol oscillator system, and its frequency is fixed at the center frequency of the i-f stages in the receiver to ensure correct operation. The keying pulse on the plate of the electronic switch, the other half of V721, places the 65-kc signal on the screen grid of mixer V722. The control grid of the mixer is being supplied with a continuous signal of about 90 kc from the unicontrol oscillator. Oil the plate of the mixer are the usual components of heterodyning. A hand-pass filter with a range of from 22 to 29 kc selects the correct component and passes it to the grid of power-output tube V723. This tube develops, at the desired level, the excitation for the transmitter intermediate power amplifier, V731 (figure 8-4) which it drives through T711. The QHB transmitter employs a pulse-type amplifier that uses three type-715C beam power tetrodes-two in parallel for the output stage and one as a driver.

The high-voltage d-c supply for the anodes of these tubes in pulse operation has a storage capacity that prevents too rapid a decay of the voltage during transmission and the rectifier tubes V729 and V730 provide a recovery rate adequate to maintain the desired operating values of d-r voltage. The storage is provided by a group of capacitors totaling 60 microfarads. These capacitors are charged to a nominal value of 3,700 volts when the equipment is in the *listen* position and therefore not keying. When the equipment is

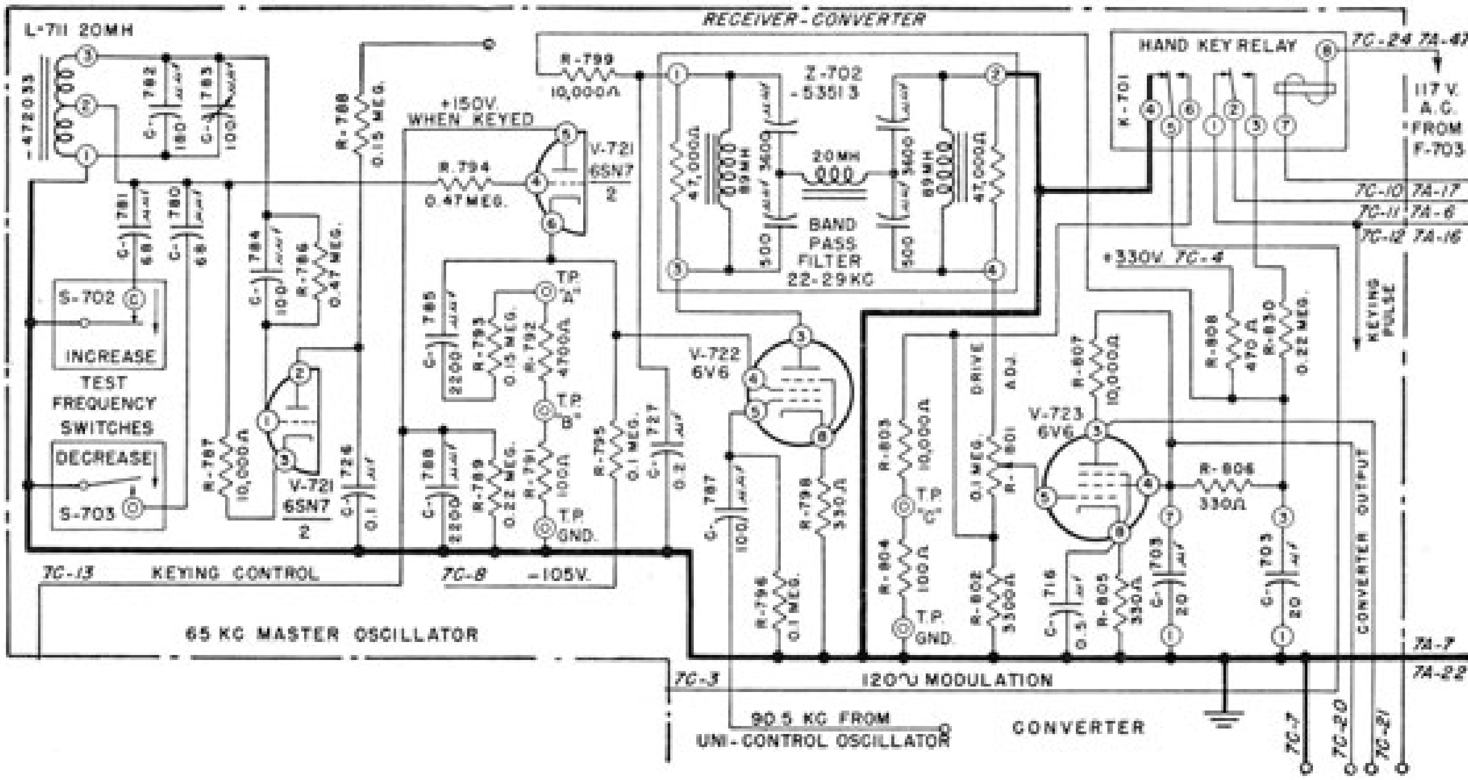


Figure 8-3.-Converter circuit.

echo ranging, the drop in d-c voltage as well as the energy delivered to the transducer during the 35millisecond transmitting period depends on the converter gain adjustment.

The value to which the d-c voltage rises during recovery depends on the keying interval and is fixed by circuit constants. With a converter gain adjustment such that the attenuation from beginning to end of the pulse is 4 db, the power, when the 3,750-yard keying interval is used, has a peak value of 7.2 kw and an average value of 6 kw.

When the converter gain adjustment is greater or less than was just mentioned then both the pulse attenuation and the power output will be respectively greater or less than the above values. With the gain adjusted for 4 db pulse attenuation, the 3,750-yard keying interval results in a d-c voltage decay of from 3,620 to 2,600 volts, whereas the 1,500-yard keying interval results in a decay of from 3,400 to 2,500 volts.

reactor, aided by the leakage reactance of the transformer, limits the primary current to a value of 9 amperes. The recovery rate is then such that, for a normal drive and a d-c voltage of 3,700 volts with no keying, a value of 3,620 volts is reached at the 3,750-yard keying interval and a value of 3,400 volts at the 1,500-yard keying interval.

A voltage pulse that is at the desired transmission frequency and that has a value of approximately 100 volts rms is produced across the primary of T711 by converter amplifier V723 (figure 8-3). The 100-volt pulse is raised to a value of 150 volts by this transformer to serve as the grid signal for driver tube V731. The plate load on this tube consists of air-core transformer T712, tuned by C800 to the center transmission frequency of the QHB equipment, and its step-down ratio provides the grid signal for the output tubes at an impedance level that ensures adequate grid drive.

The output stage, employing V732 and V733 in parallel, has as its plate load the air-core output transformer, T713, which is identical with T712 and

In order to provide a maximum rate of recovery within the peak-current limit of the rectifier tubes and to maintain a minimum power drain on the system, peak-current limiting is accomplished by an a-c reactor, L715, in the primary circuit of the rectifier transformer. Under conditions corresponding to a short circuit on the rectifier, this is tuned by C801. The secondary of this transformer supplies the transmitter output pulse to the transducer circuit at an impedance level of 50 ohms.

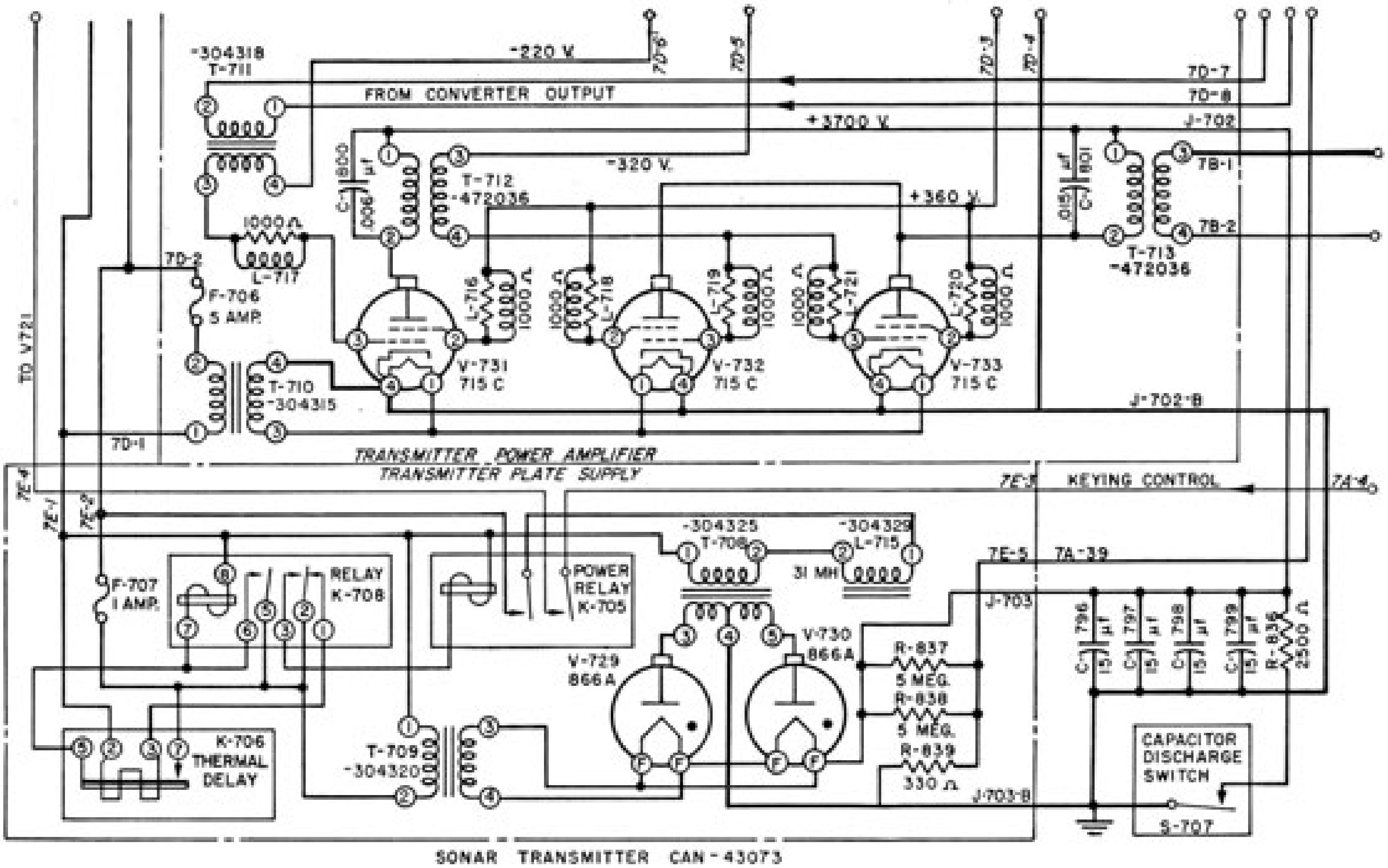


Figure 8-4.-Transmitter power-amplifier circuit.

Keying Methods

For any mode of keying, the electronic circuits that control the events must be actuated. These circuits are called *keying circuits* and may be put into action by the momentary closing of a switch, or they may be self-contained and controlled by an *RC* circuit.

These circuits and their applications are illustrated by the following discussion of the keying control circuits of the model QHB scanning sonar equipment.

KEYING CONTROL CIRCUITS

Four basic synchronized functions are performed by the keying circuits of the QHB sonar indicator control cabinet-

1. To collapse the sweep potentials of the cathode-ray indicator to start each new sweep cycle.

2. To produce a periodic adjustable-length pulse capable of controlling the transmitter output to the transducer.

3. To cause an automatic appearance of the cursor during each recycling interval.

4. To provide suitable blanking of the cathode-ray tube indicators. This function is accomplished during the recycling interval to eliminate spurious screen traces.

Three similar pulse-generating circuits of the one-shot multivibrator variety are used to produce various control pulses of definite length. Figure 8-5 shows a typical schematic of the circuit with examples of the pulses that occur at various points. The d-c electrode potentials indicated are steady-state values for the stable condition of the circuit. The duration of the generated pulse depends mainly on the time constant of C2 and R5.

Three methods of triggering these circuits are used. A positive pulse of short duration is capacitively coupled to the control grid (pin 1) of the twin triode, or a negative pulse of short duration is capacitively coupled to the control grid (pin 4). Each of these methods produces a blocking pulse

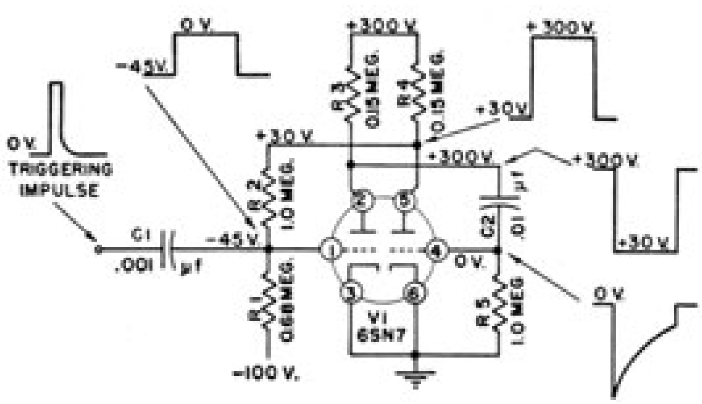


Figure 8-5 -Simplified trigger circuit.

with a duration determined by the natural period of the trigger circuit. In the third method a negative potential is coupled directly to the control grid (pin 4) of the twin triode to hold the circuit in the triggered condition as long as of this type may cause instability and undesired pulsing, particularly if the transients are steep-fronted. If regulated sources are provided for both power supplies of the trigger circuit and if an adequate bias of at least twice cut-off is used for the nonconducting triode, a stable circuit that can be positively controlled is obtained.

PRIMARY KEYING PULSE

A primary control pulse suitable for triggering the various circuits at required intervals is obtained by periodic firing of thyratron V111, a 2050-type tube (figure 8-6). Anode voltage of +150 volts is applied through the normally closed contacts (pins 1 and 2) of the sweep relay, K105, and the cathode is

desired.

In general, small transients in either the positive or the negative power supplies of a trigger circuit connected to ground through R165 and R166 in series. Firing of the tube is controlled by its grid voltage, which is derived from two sources-an automatically variable component and a calibrating component.

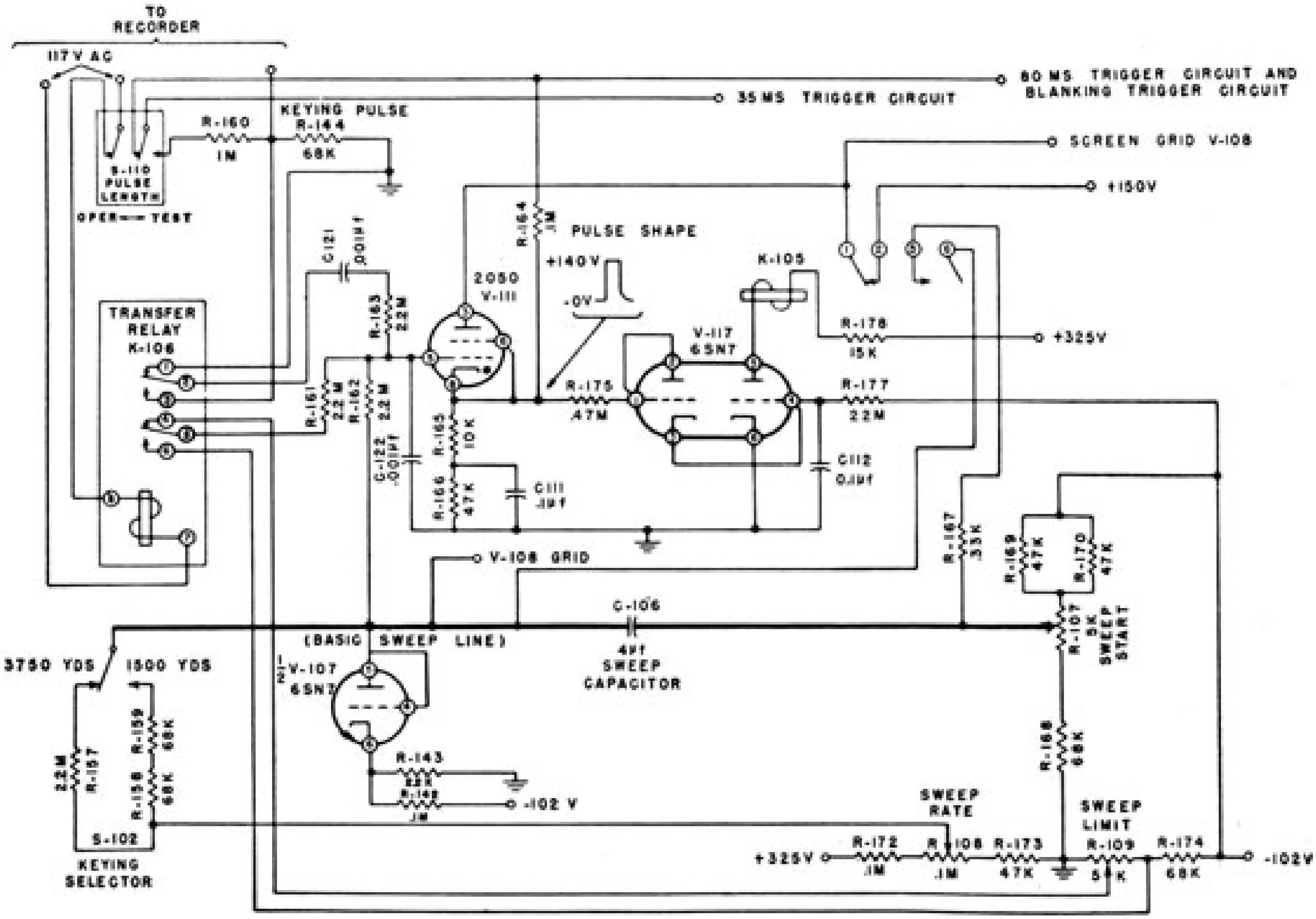


Figure 8-6. -Primary keying-pulse circuit.

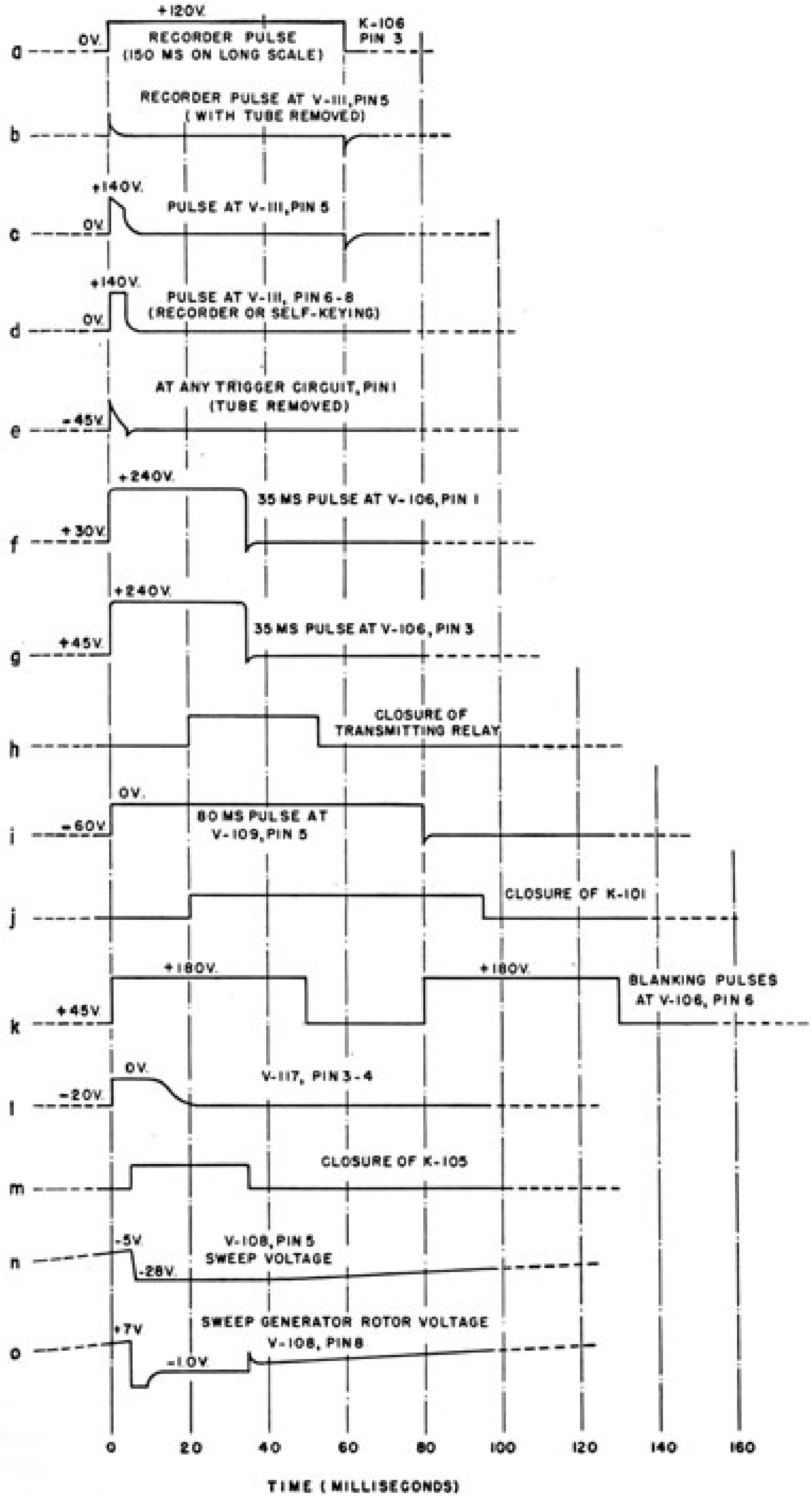


Figure 8-7 -Pulse diagrams.

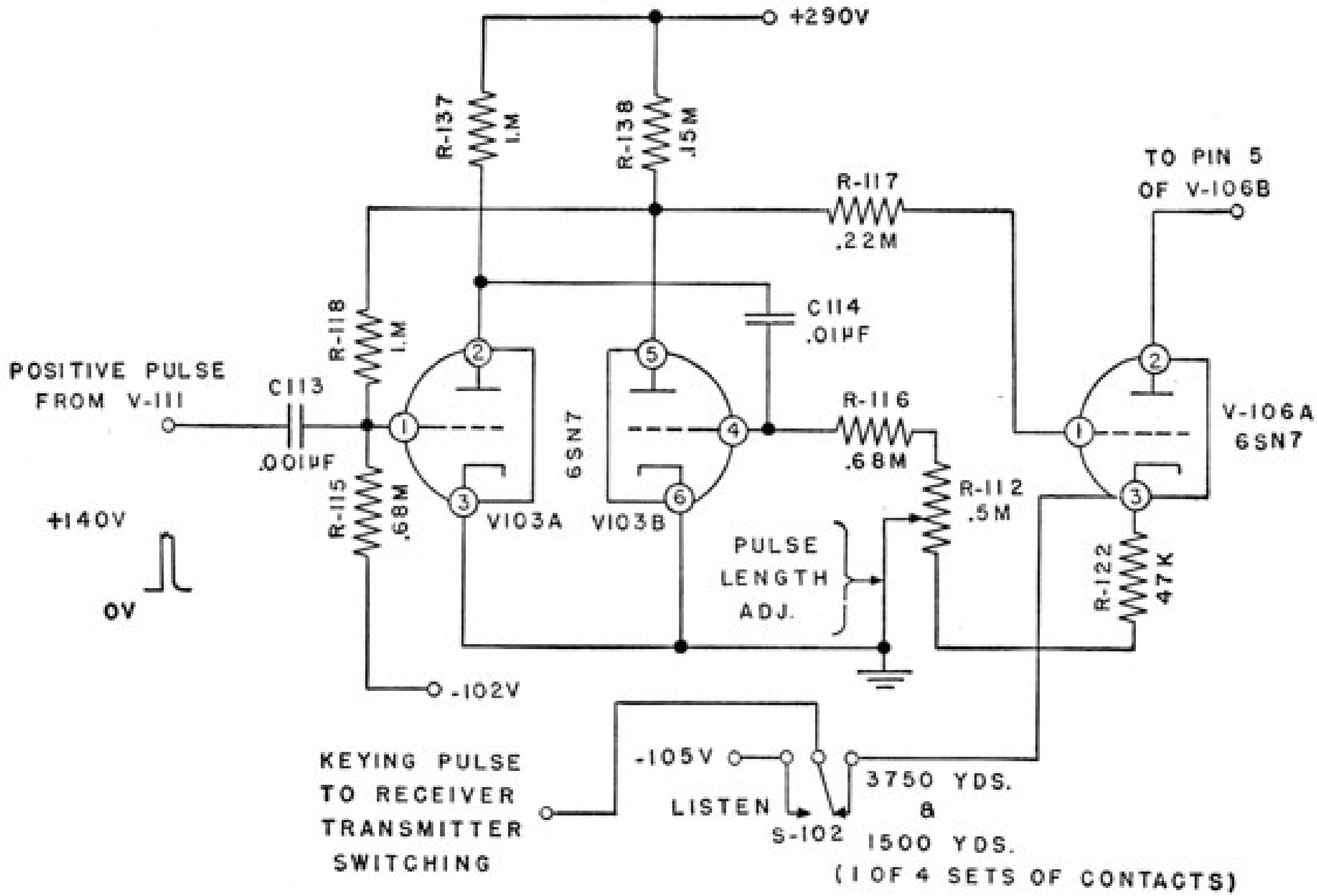


Figure 8-8. -35-millisecond trigger circuit.

The variable component is the sweep-control voltage at the sweep capacitor, C106, which swings from -28 volts to -5 volts during the sweep cycle. This potential is coupled to the grid of the pulse-generator tube, Viii, by an isolating resistor, R162. The calibrating component is derived at the sweep-limit potentiometer, R109, which is part of a voltage divider on the negative 102-volt line. The range of the calibrating voltage is from about zero to -7 volts. This voltage is connected to the control grid of the pulse-generator tube, V111. The calibrating voltage sets the firing point of the pulse-generator tube so that the sweep-voltage component corresponding to the maximum range of the screen of the cathode-ray tube causes the pulse-generator tube to conduct. The cathode and causes the triode to conduct. The 20-volt negative bias is caused by another voltage-divider network between the -102 volt line and ground that includes R177, the diode connected section of V117, R175, R165, and R166. The conduction of the triode section of V117 causes the sweep relay, K105, in its plate circuit, to operate.

Contacts *1* and *2* interrupt the anode supply of the pulse-generator tube and extinguish the tube, while contacts *5* and *6* close and discharge the sweep capacitor, C106, reducing the control-grid voltage at the pulse-generator tube to about -28 volts. The sweep relay, K105, holds in until CI12 becomes negatively charged through R177 to a magnitude such that the triode current reaches the drop-out

potential of the pulse-generator tube increases immediately from zero to about +140 volts. This voltage increase is coupled by means of a resistor, R175, and the diode-connected section of the recycling tube, V117, to the control grid (pin 4) of V117, where it overcomes the 20-volt negative bias

value for the relay. Upon being restored to normal, the sweep relay, K105, reapplies anode voltage to the pulse-generator tube, and the sweep capacitor begins its new charging cycle. diagrams *l*, *m*, and *n* of figure 8-7 show these relations.

The pulse (diagram *d*, figure 8-7) of +140 volts at the cathode of the pulse-generator tube is the

---

primary keying pulse used as a control to trigger other keying circuits. Capacitor C111 between the junction of R165 and R166 and ground, in the cathode circuit of the pulse-generator tube, delays the pulse decay slightly so that the magnitude of the negative kick is reduced when the primary pulse is capacitively coupled to the circuits it controls. Diagram e of figure 8-7, shows the triggering impulse as it appears at the control grid (pin 1) of any trigger circuit with the tube removed from its socket.

TRANSMITTER KEYING PULSE

The positive pulse that actuates the keying relay through the keying-relay tube is generated in the 35-millisecond trigger circuit (figure 8-8). The primary control pulse initiates the cycle of this one-shot multivibrator. The duration of the pulse generated in this circuit is controllable by variation of potentiometer R112. The positive pulse generated at the anode (pin 5) of the twin triode, V103, is coupled to the keying-relay circuit by way of V106A, connected as a cathode follower. This cathode follower transmits the pulse to the contacts of keying selector switch S102. When thrown to the *listen* position, the keying selector switch, 5102, connects the external keying line to -105 volts. Diagram f of figure 8-7 shows the outgoing transmitter-control pulse that causes the transmitter to operate. During the 20-millisecond pull-in time of K101

Manual production of the cursor at any time for any desired interval involves holding in the deflection-transfer relay, K101, for that interval. To hold it in, an a-c-operated relay (cursor hold), K104 interrupts the primary keying pulse and grounds the grid circuit so as to allow normal cycling of sweep and transmitter while this circuit is manually operated.

CATHODE-RAY TUBE BLANKING PULSES

Each time the transmitter is energized and while the deflection-transfer relay is closing, the cathode-ray tube must be blanked The 50-millisecond trigger circuit, V105, generates the controlling pulse for V106 which has its cathode connected directly to the cathode circuits of the cathode-ray tube. The primary keying pulse initiates the first cycle of this trigger circuit, which is a one-shot multivibrator. The operating level of the V106 cathode is determined by the intensity control (R103) setting, and the blanking pulse momentarily drives the cathode positive to cut off the electron beam in each cathode-ray tube. In diagrams *j*, *k*, and *i* of figure 8-7 the first blanking pulse starts at zero time with the other trigger pulses. The deflection-transfer relay is closed about 20 milliseconds later while the tubes are blanked. At 50 milliseconds, because the first blanking pulse is over, the cursor is allowed to appear on the screen. At the end of the 80-millisecond pulse the negative swing at the anode (pin 5) of V104 is coupled to the grid (pin 4) of V105, causing it to conduct for a second blanking

the blanking trigger circuit blanks the cathode-ray tubes.

DEFLECTION-TRANSFER KEYING OF THE CATHODE-RAY TUBE

The 80-millisecond trigger circuit generates the controlling pulse for V109, a triode-connected 6Y6, which has the coil of the deflection-transfer relay, K101, in its anode circuit (figure 8-9). The primary keying pulse initiates the cycle of this one-shot multivibrator, V104. Each time the sweep circuit is recycled the signal source for the deflection coils of the cathode-ray tube is transferred from spiral sweep to cursor deflection by operation of K101. Diagram *i* of figure 8-7 shows the pulse at the control grid of V109 (pin 5), and diagram j indicates typical operation of the deflection-transfer relay. The cursor is a radial line of variable length under the control of a range-calibrated potentiometer.

pulse, which begins immediately. The deflection-transfer relay drops out about 15 milliseconds later while the tubes are blanked, and the blanking pulse ends after its natural 50-millisecond period, thus restoring the spiral sweep for video-signal indication.

The same cycle takes place when the cursor is produced manually by operation of the cursor-bold relay except that the primary keying pulse is disconnected from the control grid (pin 1) of the blanking trigger tube, V105, which is in the 80-millisecond circuit. The first blanking pulse is caused by coupling a negative firing impulse through C120 to the control grid (pin 4) of the blanking-trigger tube. The cursor appears at the end of this first blanking pulse and stays on the screen as long as the cursor-hold relay is

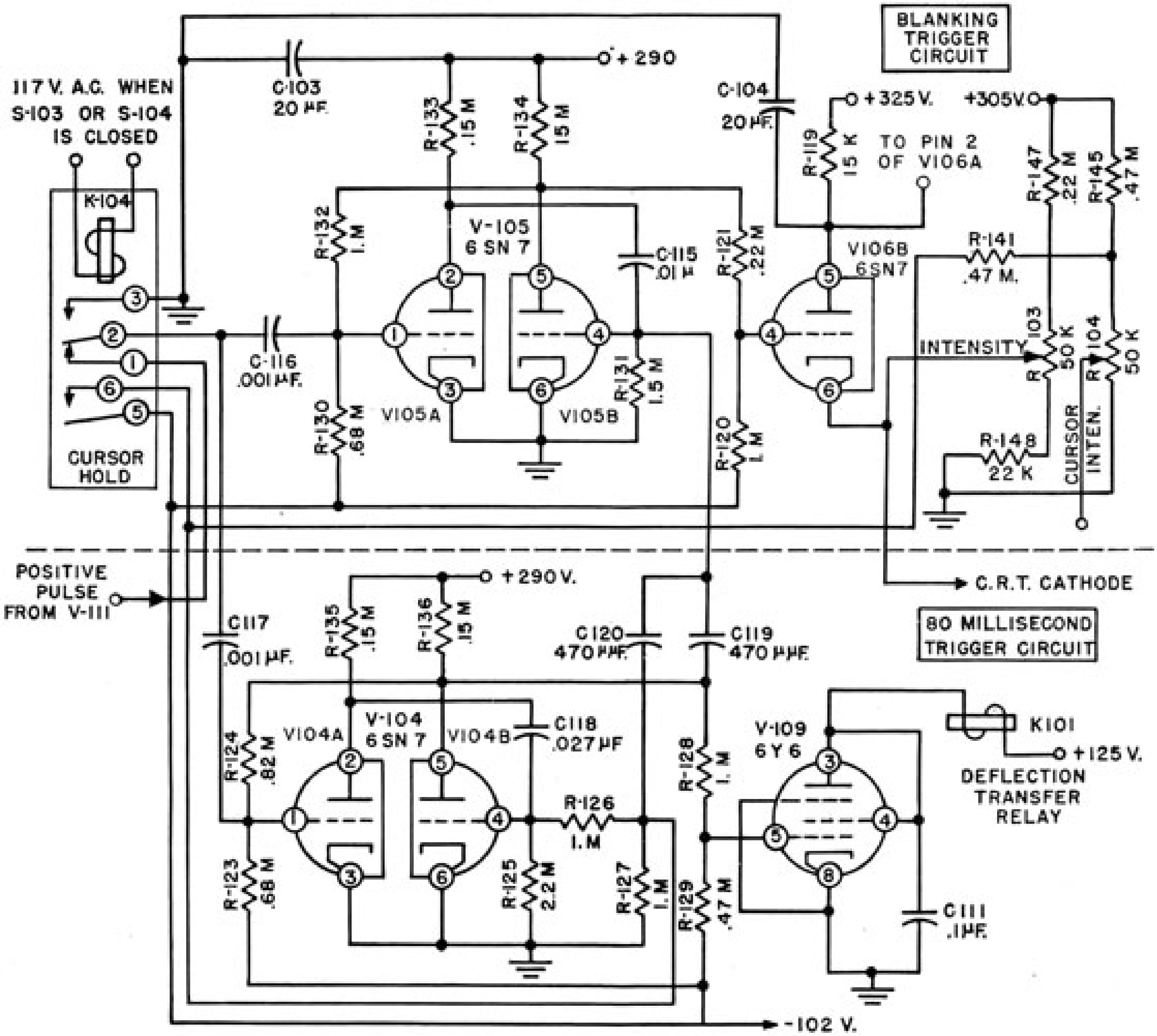


Figure 8-9. -80-millisecond and blanking trigger circuits.

energized. Upon the release of the cursor-hold relay, the 80-millisecond trigger tube is restored to normal and the negative swing at the anode is coupled to the control grid of the blanking-trigger tube to produce the second blanking pulse exactly as in automatic operation.

When the cursor is being held, the potential of -102 volts at pin 5 of the cursor-hold relay K104, is coupled to the junction of the cursor-intensity voltage divider so as to reduce the potential of the cursor-intensity control line so that the apparent

EXTERNAL KEYING

The external keying of the equipment by e range recorder involves (1) automatic operation of the a-c keying transfer relay, K106, by a single-phase potential from the external keying device and (2) coupling of its keying pulse to the control grid of the pulse-generator tube, V111, to trigger all the keying circuits (figure 8-6). The keying pulse line, connected to pin 3 of the keying transfer relay, is loaded down by R144 in order to reduce its impedance and make it somewhat less susceptible to

brilliance of the cursor is about the same when it is held as when it appears automatically for only two cycles.

transient voltages. When the keying transfer relay, K106, is operated, the external

keying pulse is capacitively coupled to the control grid of the pulse-generator tube, V111. Diagrams *a* and *b* of figure 8-7 are typical. Capacitor C122 from the grid to ground and series resistor R163 filter out high-frequency components, which tend to cause promiscuous triggering of the pulse-generator tube.

Pin 5 of the keying transfer relay, K106, transfers the grid-circuit reference potential from its normal sweep-limit value to pin 6, where a slightly greater negative value is obtained from the junction of R174 and R109. This value of sweep limit represents a sweep deflection beyond scale limits and beyond the limiting value set up in the right-hand diode section of V107, so that in the absence of an external pulse the sweep progresses to an off-screen position and stays there until the next external keying pulse.

Transmission Circuits

Transmission circuits include a converter, a power amplifier with associated power supplies, and transmit-receive switching circuits. The converter, as part of the unicontrol oscillator system, produces a voltage pulse at the frequency desired for transmission. This pulse is delivered to the transmitter power amplifier and results in the signal pulse for energizing the transducer. The transmit-receive switching involves those circuits necessary for (1) connecting the transducer to the power amplifier for transmitting, and (2) subsequently connecting the transducer to the receiver for listening. Included are the components and connections for maintenance of close contact (MCC) transmission.

For any mode of keying, the electronic switch V721B in the converter (figure 8-3) conducts by application of +150 volts to its anode upon closure of keying relay K401 (figure 8-10). V721A (figure 8-3) is the 65-kc master oscillator, which is operating at all times and which applies a voltage at this frequency to the control grid of the electronic-switch section. The electronic switch is line through a 0.1-megohm resistor. The signal developed across this resistor when the switch is conducting is tied to the screen grid of mixer V722. When the switch is not conducting, -105 volts appears on the screen grid of mixer V722 and prevents it from conducting.

During automatic keying, the pulse of +140 volts originating in the 35-millisecond trigger circuit is connected to the grid of V452, the keying-relay tube. While this pulse is incident on the grid of V452 the keying relay, K401, in its anode circuit is closed. For hand-key operation the circuit is switched so that the positive pulse for the grid of V452 is supplied by hand-key relay K701, which closes each time the hand key is closed. This relay further supplies a 120-cycle modulation to the converter power-output tube, V723. The magnitude of this modulation is such that the output is zero for an appreciable portion of each cycle, thus reducing the power dissipation in the transmitter tubes (figure 8-3). The detailed operation of the converter circuit has been explained in a preceding part of this chapter.

a cathode follower with its cathode tied to the -105-volt

Unicontrol Oscillator System

The QHB unicontrol oscillator system has been explained briefly in connection with the QHB receiver, in chapter 7. The purpose of this system is to resonate both the transmitter and the receiver to the same frequency with a single tuning control.

The principle of operation of the unicontrol oscillator system is based on the use of two independent oscillators. One of these, known as a master oscillator, is designed to operate on a fixed frequency. This frequency is the center frequency of the i-f stages. The other oscillator is the unicontrol oscillator. It is designed to tune over a band of frequencies.

The outputs of these oscillators are heterodyned in a converter, and the usual products of heterodyning are present in the converter output. A band-pass filter selects the correct frequency, which is then coupled to the intermediate power amplifier of the transmitter. The output of the unicontrol oscillator also is coupled to the mixer in the receiver.

To understand the functioning of the system, assume the following case. An echo-ranging equipment is designed to operate over a frequency band of from 22 to 29 kc with the optimum performance of the transducer at 25 kc. The i-f

---

transformers are tuned to 65 kc. With these conditions the frequency of the master oscillator is fixed at 65 kc-the frequency of the intermediate frequency-whereas the unicontrol oscillator is tunable from 87 kc to 94 kc.

When the outputs of these oscillators are heterodyned in the converter the frequencies in the output of the converter are (1) the frequencies of each oscillator, (2) the sum of their frequencies, and (3) the difference of their frequencies. As the difference frequency is the one desired, the output filter is designed to pass only those frequencies that are between 22 and 29 kc. The difference frequency at the proper power level is then delivered to the transmitter.

The frequency-response characteristics of the transducer are fixed by design. Because the equipment should operate at the optimum response frequency of the transducer, the unicontrol oscillator is tuned so that when it is heterodyned with the 65-kc master oscillator the difference frequency is at the optimum response of the transducer. Assume that the optimum frequency of the transducer is 25 kc. In this case, the unicontrol oscillator is tuned to 90 kc so that when it is heterodyned with the 65-kc master oscillator the difference frequency is 25 kc.

The output of the unicontrol oscillator is coupled also to the receiver and is present in the mixer at all times. When an echo signal at 25 kc arrives at the mixer it is heterodyned with the 90 kc from the unicontrol oscillator, and again the difference frequency is the one that is desired. The difference frequency in this case is 65 kc, which is the frequency of the i-f stages.

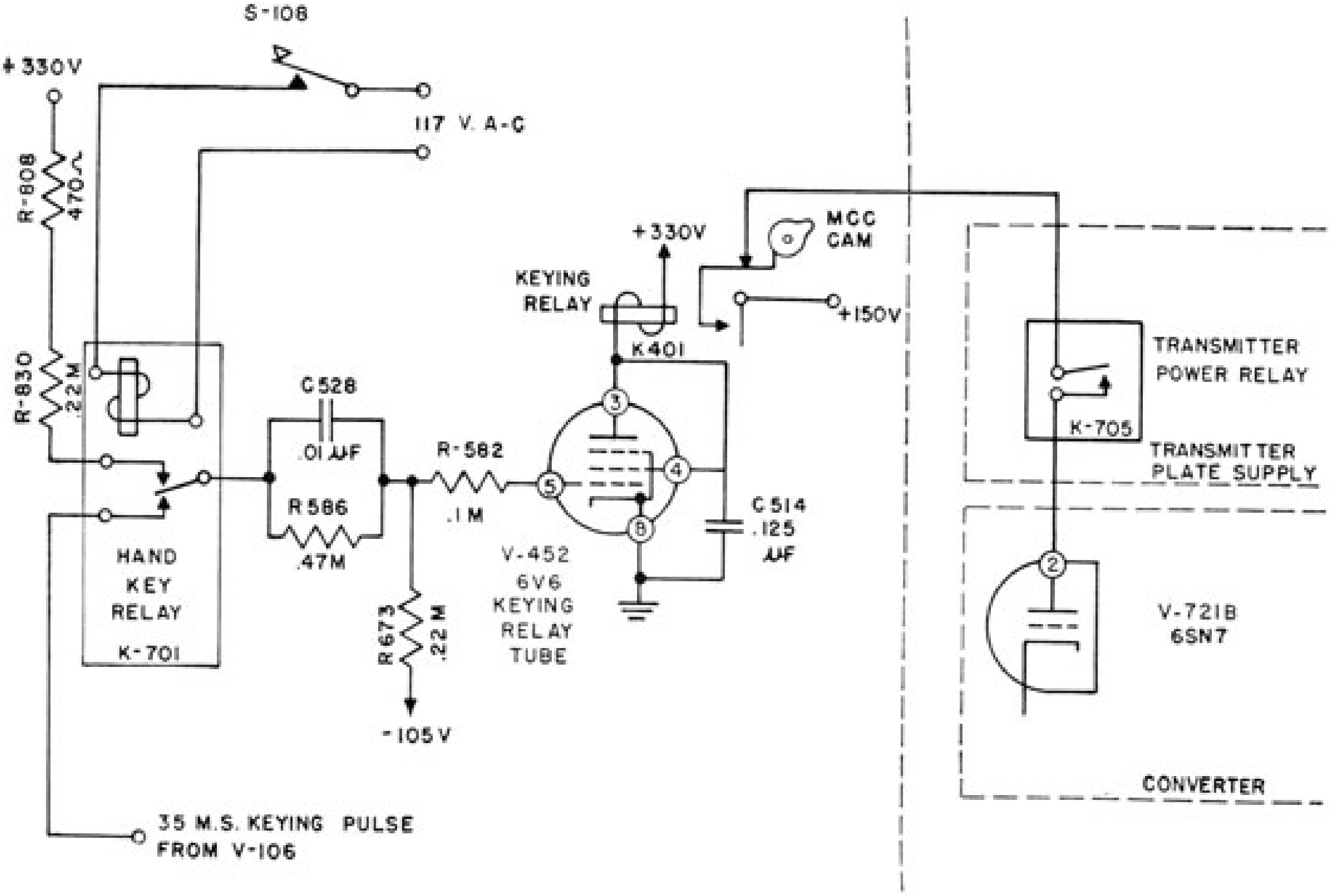


Figure 8-10. -Keying-relay circuit for the scanning-switch assembly.

CHAPTER 9

STABILIZATION

Introduction

In order to understand the theory of operation of a stabilized sonar system, it is necessary first to comprehend the general nature of the stabilization problem. A brief explanation of the stabilization problem is provided in the first part of this chapter. Following this explanation, the stabilization problem as it affects the two basic types of stabilized sonar systems is described. Readers not familiar with fire control symbols should refer to the Appendix before studying this chapter. A brief summary of the stabilizing system is given at the end of this chapter.

The Stabilization Problem

The position or direction of a line in space-whether it is a line of sight, a line of sound, or any other line-is specified by angles measured about certain reference axes the positions of which are known. In many fire control systems, the position of the line of sound is established by angles measured about axes which are horizontal and vertical in space. These horizontal, true-zenith axes are used to measure such angles as relative bearing *Brq* and depression *Eq* for the dual single-axis and three-axis systems. These systems are used in the two types of stabilized sonar equipment.

Equipment mounted on the plane of the deck, or on a plane parallel to the deck, however, positions a line in space in response to information which is ultimately referred to one of the deck systems of coordinates in which the reference axes are perpendicular and parallel to the deck. This information is called a *deck, deck-zenith system of coordinates*. These coordinates apply to such angles as sonar train *B'r'q* and depression *E'q's* for the dual single-axis system. (See figure 9-7.)

Ordinarily, the horizontal, true-zenith axes and the deck, deck-zenith axes are in alignment only at the infrequent intervals when the deck plane is horizontal. Most of the time the two sets of axes are being continuously displaced with respect to each other by the tilting movement of the deck.

In order to measure the degree of displacement of these axes from each other, a stable element is used. The stable element is normally trained to *B'r*, the bearing of the main battery director. Thus the stable element measures these angles using *B'r* as a reference. In order to stabilize the sonar equipment the reference line about which the angles are measured must be changed from *B'r* to the bearing of the sonar target. This conversion is the function of the stabilization computer.

The method by which a stabilization computer, in conjunction with a stable element, is used to relate information measured with respect to two different systems of coordinates for the purpose of stabilizing a sonar system is illustrated by the following problem. Suppose a ship is riding at anchor with the sonar equipment trained on a stationary target. As determined by the sonar operator, the line of sound is fixed in space and the angles of relative sonar bearing *Brq* and depression *Eq*, which establish the position of the line, have fixed values because they are measured with respect to a system of horizontal, true-zenith coordinates. But the position of the line of sound in the dual-single axis system is established by the sonar train *B'rq* and the depression *E'q'*, which are measured with respect to a deck, deck-zenith system of coordinates. Because the deck,

deck-zenith axes are continually moving in space as the ship rolls and pitches, these angles are continually changing in value. If the target moves, the changes in the values of these angles also become a function of the movement of the target.

To determine the varying values of *B'rq* and *E'q'* in this problem it is necessary to relate mathematically the fixed values of *Brq* and *Eq* with the continually varying angles that measure the roll and pitch of the ship. The fixed values of *Brq* and *Eq* are supplied to the stabilization computer from the sonar equipment, and the continually varying values of the angles measuring the roll and pitch of the ship are supplied to the stabilization computer from a stable element. The stabilization computer combines the trigonometric functions of these angles into the proper mathematical equations from which the continuously varying values of *B'rq* and *E'q'* can be computed. The computed values of these angles are then applied to the drive mechanisms of the system to keep it continuously positioned on the desired line of sound.

Thus the function of the computer is to receive angles measured with respect to one set of coordinate axes and the angles of roll and pitch (or angles convertible to roll and pitch) and to compute from this data angles measured with respect to another set of coordinate axes.

The stable element is the piece of equipment which provides a fixed system of axes (to which the variable angles caused by roll and pitch may be compared)-and the gyroscope is the heart of the stable element. It is important therefore to understand the characteristics of a gyroscope.

Fundamentals of the Stable Element

PROPERTIES OF A FREE GYRO

If a heavy wheel is mounted so that its shaft is free to turn in any direction, it is known as a *free gyroscope or free gyro*. Usually the mount is constructed with three mutually perpendicular axes about which the wheel may turn. Thus in figure 9-1 the wheel is free to spin about axis A,

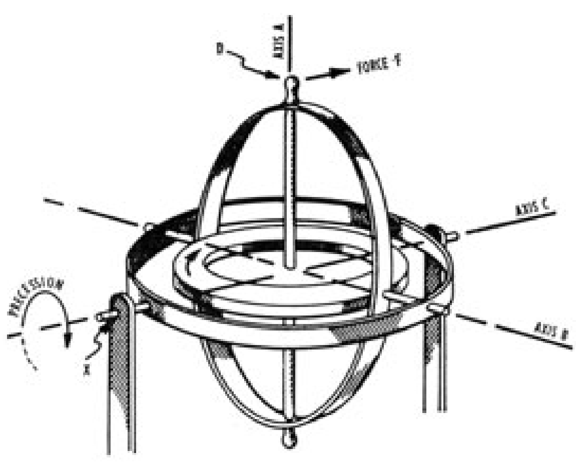


Figure 9-1 -Properties of a free gyro.

to turn about *B*, and to turn about *C*. The gyro wheel is located so that its center of mass coincides with the intersection of these three axes.

For purposes of illustration, all bearings are considered to be without friction. It is evident that the gyro wheel, when not rotating, is in a state of indifferent or neutral equilibrium; that is, it remains in any position in which it is placed. In addition, it yields in the direction of any force which tends to rotate it about one of its axes, just as any free mass moves in the direction of an applied force.

If the wheel is set spinning rapidly, it exhibits entirely new phenomena. It resists rather than yields to an applied force. A force, *F*, applied at point *D* of figure 9-1, produces a torque about the axis *B*. This torque, instead of rotating the frame in the direction of the applied force as it would do if the wheel were not spinning, is opposed by the frame.

Additionally, the wheel starts to rotate slowly (precess) about axis *C* in the direction indicated. If the mount is without friction, as was assumed, this action continues as long as the force is applied at *D.*

Similarly, a force applied at *D* in a plane through axis *B* tends to rotate the wheel about the *C* axis but the gyro resists this motion and turns instead about the axis *B*.

A pressure applied to the gyro wheel frame always results in reaction forces at the bearings. If, as in the cases illustrated by this figure, the

applied force and bearing reaction are not in the same straight line, these forces form a couple which tends to rotate the gyro wheel axis. The free gyro does not, however, turn about the couple axis but rotates about another axis perpendicular to the couple axis. Thus, in figure 9-1, a couple about axis *B* results in a rotation or precession of the wheel and frame about *C*.

Experiments show that, neglecting inertia, the gyro does not resist translation, that is, motion which keeps the spin axis A parallel to its original position.

RIGIDITY OF PLANE

Rigidity of plane is that property of a gyro by which the gyro tends to maintain the plane of its wheel parallel to its original position in space. This property results from the fact that a mass in motion can have its direction of movement changed only by applying a force to the mass.

PRECESSION

Precession is the name given to the slow movement of a gyro wheel resulting from the application of an external force or couple which tends to rotate the spin axis of the gyro.

Figure 9-2 shows a rapidly spinning gyro in which the axis of spin is *A*. A couple represented by forces *F-F'* tends to twist the gyro wheel about the couple axis *B* perpendicular to, and in the same horizontal plane as, *A*. Consider a small section of the wheel rim at *D*. Due to the rotation of the wheel, section *D* has a high linear velocity in the direction *DE*.

The couple *F-F'* exerts a force upon this small mass along *DH* and so accelerates it in this direction. During a short interval of time this acceleration gives the particle a component of velocity *DH*. The result of combining velocities *DE* and *DH* is a new velocity *DG*, equivalent to a rotation through an angle about axis *C*. Therefore the effect of a couple *F-F'* acting about the *B* axis is to cause a slow rotation of the gyro wheel about the *C* axis. This rotation is known as precession.

In order to obtain a high rigidity of plane and slow

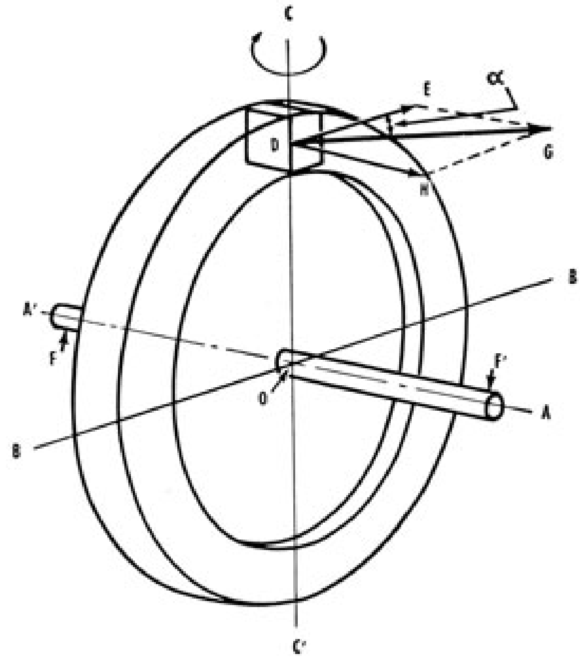


Figure 9-2 -Precession.

not turn about an axis *C* at right angles to the axis of the applied couple.

To determine the direction of precession apply the following rule: The axis of a freely mounted gyro tends to turn or precess in such a direction that it becomes parallel to the axis of the applied torque, by the shortest path, and with the rotation of the wheel in the same direction as the applied torque.

APPARENT ROTATION

Assume now that the gyro wheel supported by its universal mounting as before is placed at the equator of the earth with its *A* axis vertical, as shown in position *1* of figure 9-3. To an observer standing on the earth the wheel appears to rotate at the rate of one complete turn in 24 hours. This rotation might seem puzzling were it not remembered that it is the earth that is turning-not the gyro.

EFFECT OF FRICTION

In the practical construction of the mounting described, some friction is inevitably present at the trunnion bearings. Assuming for the moment that the bearings of the horizontal axis *B* (figure 9-1) have slight friction, it is apparent that the

precession, gyro wheels are made heavy in weight and are operated at a high rate.

It has been shown that if the gyro wheel is freely supported as in figure 9-1 and a force or couple is applied about the axis *B*, the wheel does

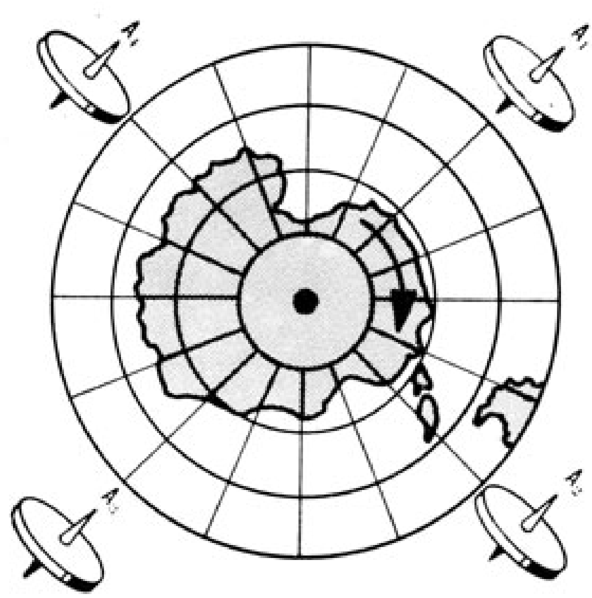


Figure 9-3 -Apparent rotation.

earth's rotation applies a slight turning moment or couple to the gyro wheel. A free gyro, however, does not turn in the direction of an applied couple but precesses around an axis 90° from the couple. Consequently the slight friction in bearings of axis *B* causes the gyro wheel to precess about the axis *C*. With proper construction of bearings *B* this precession may be made very slow.

In the case just described it was assumed that the bearings determine the ability of a gyro wheel to maintain its plane or rotation fixed in space against the friction of bearings *B*. If, to take an extreme case, the supporting frame were locked about axis *C* (X, figure 9-1), the gyro wheel would immediately lose its resistance to the friction of bearings *B* and so would partake of the earth's rotation just as if the wheel were not spinning. The importance of extreme freedom about axis *C* is therefore apparent.

EFFECT OF LATITUDE

It has been noted that the earth's rotation causes an apparent rotation of a gyro which is set spinning with its *A* axis perpendicular to the earth's surface. At the equator this apparent rotation appears to be a straight backward gyration (with respect to the earth's rotation) at the rate of one revolution every 24 hours about a north-south axis. At either pole this phenomenon does not occur (again assuming frictionless bearings), because the gyro axis *A* is already parallel to (or an extension of) the earth's axis, as in figure 9-4. At any point or latitude between the pole and the equator, however, the wheel appears to gyrate once every 24 hours about an axis parallel to the axis of the earth's rotation, and in a direction opposite to that of the earth's rotation, as in figure 9-5.

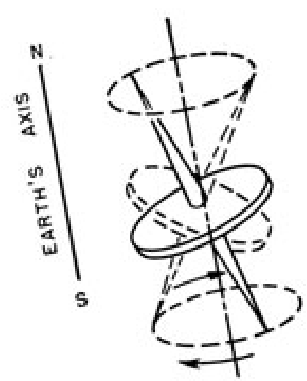


Figure 9-5 -Gyration between equator and pole.

COMPENSATION

In any application of the gyro to precision instruments, corrections for the earth's rotation, friction, acceleration, turning, and other factors must be applied if the gyro is always to spin in a fixed

Figure 9-4. -Effect of latitude.

plane with respect to the earth's surface at any latitude.

Compensations for these errors are not dealt with here, as their operation is not essential in understanding the principles of the stable element.

Stable-Element Construction

The stable element is used (1) to measure movement of the deck in level and crosslevel angles or in roll and pitch angles, depending on the connections of the stable element, and (2) to transmit these angles as synchro signals. The principal part of the stable element is an electrically driven gyroscope that establishes a horizontal-reference plane from which the level and crosslevel angles are measured.

There are three follow-up systems in stable-element equipments-the train, the crosslevel, and the level follow-up systems. In some equipments the train is locked on zero, and the outputs are in terms of roll and pitch instead of level and cross-level. The follow-up system for train determines the error between the bearing of the equipment that is being stabilized and the bearing of the training frame in the stable element. When the stable element supplies roll and pitch, however, the train follow-up system is not used and the *B'r* input to the stable element is locked in the zero position. In stable elements having outputs of level and crosslevel, the train follow-up is used to rotate the stable element to the bearing of the equipment that is being stabilized.

The follow-up systems for level and crosslevel are identical and are actuated by electric error signals, which originate in the gyro unit. These signals are amplified and are used for actuating the level and crosslevel motors, which drive not only the synchros transmitting the level and crosslevel angles to the equipment to be stabilized but also the level and crosslevel follow-up circuits. If the stable element is modified and the train input locked, the output is then roll and pitch instead of level and crosslevel.

Level and Cross-Level Receiver System

A spherical diagram illustrating the relationship between the angles of level *L*, crosslevel *Zd*, director train *B'r* roll *M*, and pitch *N* is shown in figure 9-6.

Angular inputs which indicate the attitude of the deck with respect to the horizontal are supplied to the computer by a stable element. The space reference of the stable element is a gyroscope which rotates about a vertical axis to maintain continuously an a-c electromagnet in a fixed position. Above the electromagnet are two sets of follow-up coils, the fields of which are at right

One of the stable elements from which the computer may receive tilt angle inputs is the Stable Element Mk 6. This instrument is trainable and is used usually in conjunction with the Gun Director Mk 37. As the director trains to position a line of sight, it generates a signal corresponding to the angle of director train, *B'r*, which is transmitted by synchro to the stable element in order to drive the stable element through the same angle. Because the stable element normally is trainable, its cross-level and level axes are rotated about an axis perpendicular to the deck and in a plane parallel to the deck. Pitching and rolling which may occur at the angle of director

angles to each other, one for level *L* and the other for crosslevel *Zd* control. Both sets of coils are supported on the inner gimbal of two mutually perpendicular gimbals. When the motion of the ship displaces the coils with relation to the electromagnet, follow-up systems are actuated by the coils to move the gimbals in such a direction as to restore the original position of the coils with respect to the electromagnet. The angular movement of the level and crosslevel follow-up controls causes the signals across the level *L* and crosslevel *Zd* transmitters to change in correspondence with the attitude of the gimbals and thus measure the attitude of the deck with respect to the horizontal.

239276°-53-13

train are measured about the level and crosslevel axes, respectively. Thus as the stable element trains through *B'r*, continuously changing values of *L* and *Zd* are generated for continuously varying values of *B'r* and are transmitted to the various stabilized equipments on the vessel. When the necessary electrical connections are made the same angular values are received by the computer.

If the tilt angles transmitted from a stable element, not locked on zero, were used directly in making the stabilization computations for the various fire control systems serviced by the computer,

the train angle at which these tilt angles were measured would also be required in the computations, thus needlessly complicating the computer circuits. If tilt angles measured at the fore and aft axis of the ship were used, where the angle of train is zero, only the tilt angles would be required for the stabilization computations and the computer circuits would become less complicated. For this reason the functions of *L*, *Zd*, and *B'r* are converted by the required trigonometric equations

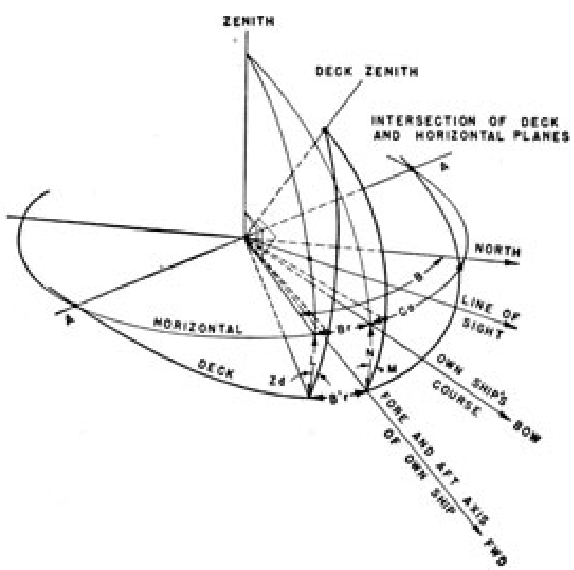


Figure 9-6. -Spherical diagram illustrating various angles in sonar problems.

into the functions of *N* and *M*, which are the tilt angles at the fore-and-aft axis.

The level and crosslevel receiver system performs this conversion. This system consists of three input servos which are positioned through the angles of level *L*, crosslevel *Zd*, and director train *B'r*. The resolvers in these servos are connected in such a manner that the components of these angular functions are related according to the equations in the functional diagrams shown in figure 9-11. The rotation of the servomechanisms accomplishes a continuous simultaneous solution of these equations. The solution is in terms of the angles of roll *M* and pitch *N*. If a gyro which has been locked on zero is used, the angles supplied to the computer are measured directly in roll and pitch, eliminating the need for the level and cross-level receiver.

ROLL AND PITCH COMPUTER SYSTEM

The computer circuit, described in the section on the level and crosslevel receiver, is located partly in the level and crosslevel receiver and partly in the roll and pitch computer. The two resolvers in that part of the computer circuit which is located in the roll and pitch computer rack control two output servos, which, in turn, position various resolvers through the angles of roll *M* and pitch *N*.

Dual Single-Axis Stabilization System

In the problem of determining a line of sound to an underwater target with a dual single-axis below-decks system like that of the QHB and QDA, two sound heads rotatable about axes in the deck must be positioned properly as shown in figures 9-7 and 9-8.

Each sound head generates along a plane of sound. The sound heads are rotated so as to make their planes of sound intersect on the line of sound to the target. Sound plane *A* is rotated about an axis perpendicular to the deck through the train angle *B'r'q* and sound plane *B* is rotated about an athwartship axis parallel to the deck through the tilt angle *E'q's*. Both of these angles are measured with respect to deck, deck-zenith coordinate axes, but the required fire control information is computed as relative bearing *Brq* and the depression angle *Eq*, both measured with respect to horizontal true-zenith coordinate axes. In order to position these sound heads properly, the computed data *Brq* and *Eq* must be converted through the use of *M* and *N* into *B'r'q* and E*'q's* in the dual single-axis stabilization system and transmitted in that form to the drive mechanisms of the two sound heads.

Note that the spherical diagram indicates the depression angle at the line of sound as *E'q'*. If this were a two-axis system, *E'q'* would be the proper depression order, but because of the QDA sound head is not trainable, in the QHB-QDA dual single-axis system, *E'q's* must be used instead.

The dual single-axis stabilization unit in the computer consists of (1) two input servos which are used to position mechanically two resolvers

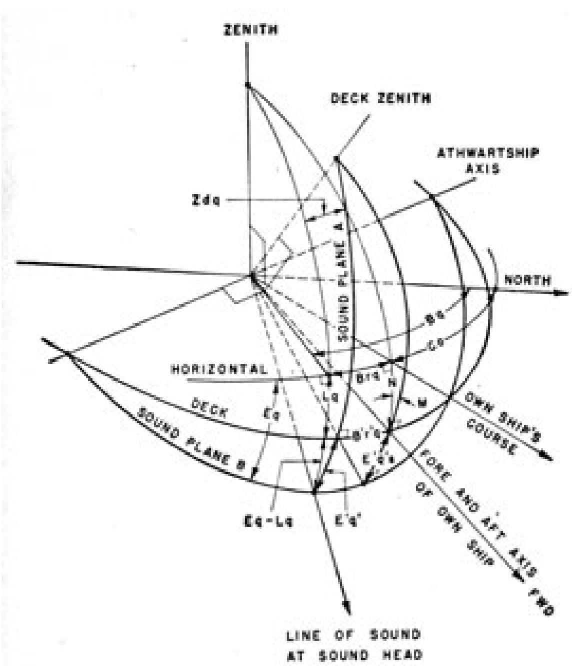


Figure 9-7 -Dual single-axis stabilization problem.

through the angles *Eq* and *Brq*, (2) two resolvers which are mechanically positioned through the angles *M* and *N* by the servos in the roll and pitch computer or the roll and pitch receiver, and (3) two servos which are positioned by their resolvers through the angles *B'r'q* and *E'q's*. The resolvers are connected in such a manner that they relate the functions of the four input angles in trigonometric equations which produce in simultaneous solution the functions of *B'r'q* and *E'q's* for synchro transmission to the drive mechanisms of the sound gear.

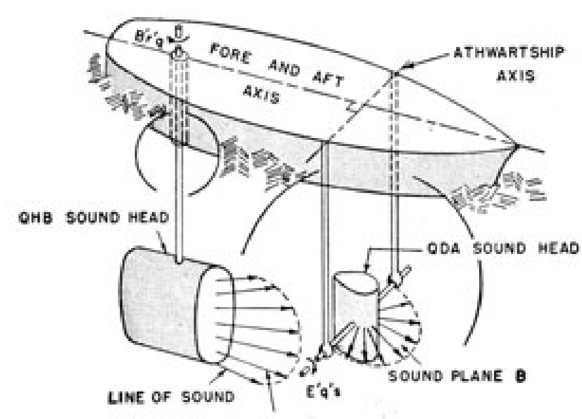


Figure 9-8 -Dual single-axis system.

Three-Axis Stabilization System

In a three-axis below-decks system a line of sound is positioned about three axes of rotation relative to the deck, as shown in figures 9-9 and 9-10.

Although movement about the three axes of rotation occurs simultaneously, for the sake of explanation, the operation of the system may be treated as though it occurred in sequence as follows. The system must train in the plane of the deck to position its crosslevel axis in a vertical plane through the line of sound, but the train angle input is computed in the horizontal-true zenith angle of relative bearing, *Brq*. Therefore, this angle must be converted by parts of the computer circuit in the three-axis stabilization system through the angles of roll *M* and pitch *N* into the deck, deck-zenith angle of *B'rq*. Once it has been positioned in the vertical plane through the line of sound, the crosslevel axis is rotated through the crosslevel angle *Zdq* until the level axis is horizontal. Then the level gimbal is rotated about the level axis through the level angle *Lq* to bring the sound head into the horizontal plane. From this point the level gimbal is tilted through the depression angle *Eq* to bring the sound head onto the desired line of sound. In actual practice, the latter two steps are accomplished by continuously subtracting *Lq* from *Eq*. The angles *Zdq* and *Lq* are derived by the simultaneous solution of trigonometric equations relating *M*, and *N*, and *Brq*.

The three-axis stabilization system utilizes two sections in the Mk 59 computer. In one section, resolvers are mechanically positioned by input servos through the angles *M*, *N*, and *Brq*. The resolvers are connected so that they solve for *Zdq* and *Lq*, which are then introduced by means of their respective output servos into the stabilizing drive mechanisms of a three-axis below-decks system like the SQG sonar. The synchro differential transmitters in the *Lq* output servo receive *Lq* as a mechanical displacement and subtract it

191

from *Eq*, which is received as an electrical signal. In the other section, *Brq*, *N*, and *M* are introduced into their resolvers in that order to produce the train angle *B'rq*. An output servo positions the drive mechanism of the sonar gear through *B'rq*.

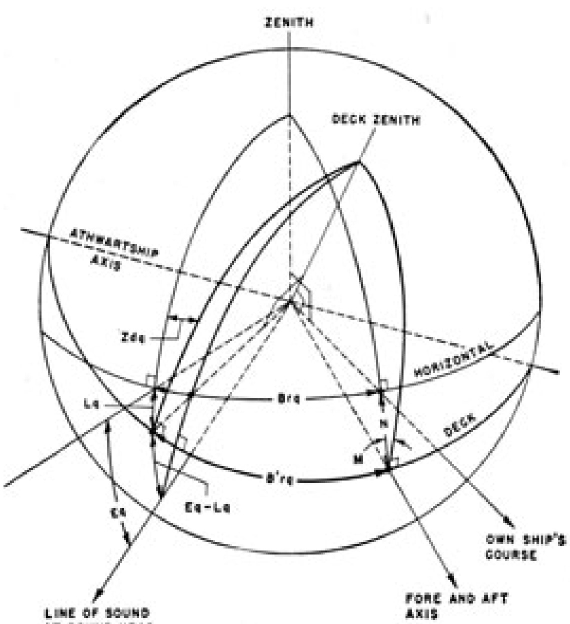


Figure 9-9 -Three-axis stabilization system problem.

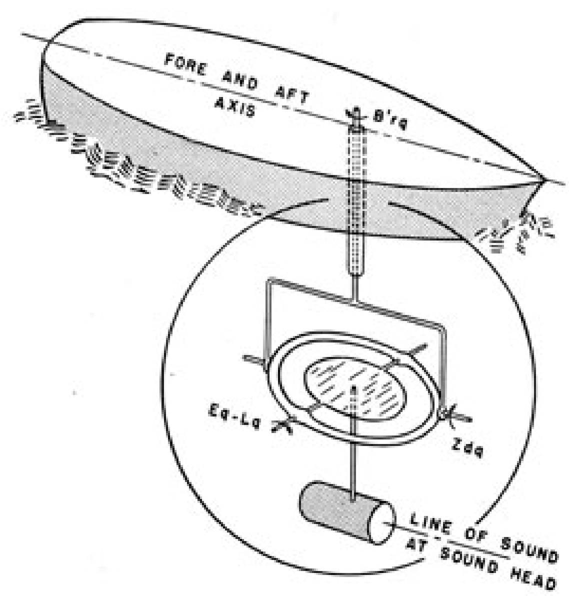


Figure 9-10 -Three-axis system.

Computer Units

The functional diagram in figure 9-11 shows the complete computer with 5 stabilized outputs. The number of outputs can be altered to supply the needs of different installations. In determining the composition of a Computer Mk 59, the following factors are considered:

1. The type of stable element used to furnish deck tilt angle data. If the stable element is the Mk 6 or an equivalent, a level and crosslevel receiver rack and a roll and pitch computer rack are required. If the stable element is the Mk 7 with alteration for roll and pitch output, or an equivalent, only a roll and pitch receiver is required.

2. The types and quantities of stabilization racks required are directly dependent on the types and quantities of fire control gear on board ship which require stabilization.

3. The inclusion of a power supply with the computer depends upon the requirements of the particular fire control system with which the computer is to be associated. If the fire control system operates off a central supply, the power supply rack may be omitted from the computer. If the fire control system operates off a series of decentralized power supplies, a power supply for the computer may be necessary. The use of a power supply determines the number and size of terminal compartments required. If a power supply is used, a single large compartment is usually sufficient. If no power supply is used, it is more convenient for symmetrical arrangement to use two smaller terminal compartments.

Integrated Sonar System

The stabilization computer and stable element are ordnance equipments and therefore are not among the equipments maintained by the electronics division. The information in this chapter is included so that the electronics officer can understand better the nature of the electrical orders of

---

FOLDOUT - Figure 9-11 -Functional diagram, Mk 59 Mod 0.

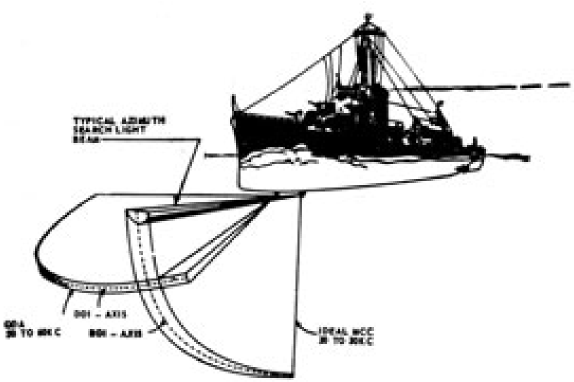


Figure 9-12 -Typical QDA, normal azimuth, and MCC azimuth sonar beams.

stabilization injected into the sonar equipments.

Figure 9-12 shows the sound beams used in a complete stabilized sonar system. In practice, the edges of the beams are not sharp lines as shown. The normal azimuth beam, shown in figure 9-12 as a half-round pencil, is employed during search and while closing the target. If the target is deep, the normal azimuth beam passes above it except at long ranges. Thus the operator can use the *maintenance of close contact* (MCC) feature for the form of level and crosslevel angles, and, therefore, the computer is equipped with a level and crosslevel receiver, which must first convert these angles to roll and pitch in order for them to be used in the computer. Note the *by-pass switch* which, when thrown to the *search* position, allows the azimuth sonar to receive stabilization signals, but the depth-determining equipment is unstabilized. In the *attack* position both of the equipments are stabilized.

The principal functions of the stabilization computer are:

1. To convert the angles of level, *L*, and cross-level, *Zd*, which exist when the stable element is trained away from 0 relative bearing, to angles of roll, *M*, and pitch, *N*. The information obtained from this conversion can be used to stabilize the sonar system, which is not usually on the same bearing as stable element train, *B'r*. In some systems the stable element has its *B'r* input locked on zero. In this case its output is in roll, *M*, and pitch, *N*, directly, thus eliminating the need for this function of the

target s at close range. The MCC beam used in this system is approximately the same width in the horizontal direction as the normal azimuth beam, but the MCC beam extends over most of the quadrant from the surface downward. Although the MCC beam is inefficient for use at long ranges, at short ranges the echo strength increases sufficiently to permit the azimuth equipment to maintain contact and to determine range and bearing practically up to the point of passing over the target.

Figure 9-13 illustrates the interflow of functions between the units comprising a complete stabilized sonar system. The system illustrated comprises the QDA, QHB, and OKA, and is one of the systems in use today. The input to the system is in computer.

2. To combine roll and pitch with the relative sonar target bearing in the horizontal plane *Brq*, to produce *B'r'q*, the relative bearing in the deck plane, in order to maintain the azimuth transducer on the same true bearing and hence counteract the tendency for roll and pitch to carry the beam to one side or the other.

3. To combine roll, pitch, and relative sonar target bearing, *Brq*, with the depression of the beam relative to the horizon, in order to compute E'q', the beam depression relative to the deck.

The order, *E'q'*, is the correct tilt angle for all phases of ship roll and pitch of a depth transducer which is trained on the target, as is the case in a three-axis system. *E'q'* is also the correct depression of that portion of the broad QDA beam which extends along the bearing of the target.

Tangent Solver

If the target lies dead ahead, the order, *E'q'*, is the correct order to which to tilt the QDA transducer to contact the target for any phase of ship roll or pitch. If the target bears off the bow, however, the QDA transducer must be tilted to a somewhat greater angle than *E'q'* in order to contact the target because for any position of the transducer, the beam depression is greatest dead ahead, and is less for bearings off the bow. If the beam were broad enough, the depression of the portions of the beam that extend athwartships would be zero; that is, the beam would lie along the

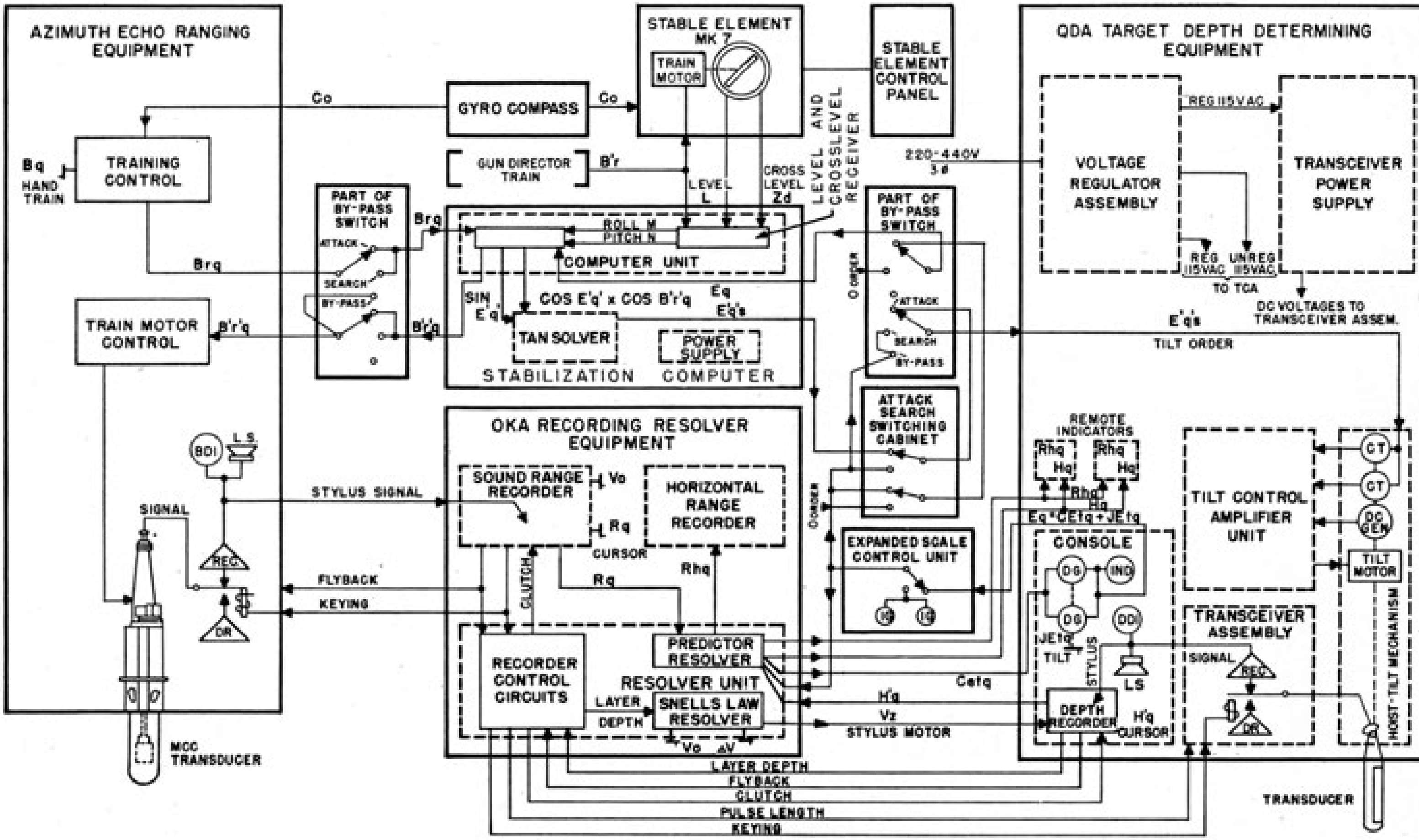


Figure 9-13. -Block diagram of a stabilized sonar system.

surface at 090 and 270 relative. If the transducer were tilted to 10°, the beam in the forward direction would be depressed 10°, while the portion of the beam extending along a bearing of 045 relative would be depressed only about 7°.

The tilt order, *E'q's*, produced in the tangent solver is conveyed through the attack search switching cabinet and by-pass switch to the CT synchros which are mechanically geared to the shaft of the hoist-tilt mechanism in the QDA. If the transducer is not in the position represented by the order *E'q's*, the rotors of the CT synchros will not be oriented properly with respect to the fields produced by the order signal. As a result, a voltage is delivered by the CT synchro rotors to the tilt-control amplifier unit, which drive the system in the proper direction to align it with the order, *E'q's*.

Conclusion

In short, the heart of the stabilized sonar system is the computer, which receives stabilizing inputs from the stable element. The stable element uses either the main director train or the fore-and-aft axis of the ship as its reference. The computer converts these stabilizing inputs into the proper voltages to be used with the stabilized sonar system, which is usually on a different reference axis. The computer also supplies stabilizing information to different units of equipment, such as searchlights, radar systems, torpedo tubes, and guns, as required by the ship.

As the ship rolls and pitches the stable element measures the angles of deviation from the true horizontal. The computer then converts these angles into level and crosslevel signals to be added to the angles of depression and train, which are generated in the sonar system. The result is that the lines or planes of sound remain stable and on the target regardless of the gyrations of the ship.

195

[Previous Chapter](#) [Sonar Home Page](#) [Next Chapter](#)

CHAPTER 13

SUBMARINE LISTENING EQUIPMENT

Introduction

Submarine listening equipment is designed to receive and reproduce underwater sounds-both sonic and ultrasonic-for the purpose of identifying the sounds and locating their sources. Sonic sounds (below 15,000 cycles per second) are made by propellers, engines, rudder motors, pumps, gear wheels, and many other devices. Ultrasonic sounds originate mostly from high-speed propellers. The bearings of the sources of sounds usually can be determined, so that targets can be located without the use of echo-ranging gear.

The original J-series listening equipment was designed for use on submarines. Most modern listening equipment, such as the JP and JT, is designed for patrol craft, picket boats, and submarines. The JP-series listening equipment is now in use on submarines as a unit of the JT equipment.

The JP is a sonic equipment-that is, it receives audible sounds, amplifies them, and applies them to either headphones, loudspeakers, or a tuning-eye indicator. Because the line hydrophone used with the JP is moderately directional, bearings on the sound sources can be made by use of the tuning-eye indicator. The JP equipment was designed for small surface craft. The JP-1, JP-2, and JP-3 are used on submarines.

Although the JP is a complete sonic listening equipment, it is now used on submarines only as a part of the JT equipment. The JT equipment uses a directional line hydrophone to receive both sonic and ultrasonic noises. The JT uses the JP amplifier and indicator practically unchanged. In addition, the JT has (1) a beat-frequency converter for converting ultrasonic sounds into audible frequencies and (2) a right-left indicator for taking accurate bearings on sonic sounds. The JP and JT equipments are described in this chapter, as is the JAA triangulation-listening-ranging equipment.

The JAA equipment consists of two line-type hydrophones and their associated amplifiers. One hydrophone is mounted on the forward end of the submarine and the other on the after end. Either hydrophone can be used independently to locate targets by listening, or both hydrophones can be used simultaneously on one target. When both are used, the range of the target can be calculated by triangulation of the sound emitted from the target vessel. The JAA bearing-indicating units are similar to those of the JT.

This chapter discusses not only the JP, JT, and JAA listening equipments but also the following accessories to submarine listening equipments: (1) The noise-level monitor and cavitation indicator, which checks the noise level and the cavitation noise originating from own ship; and (2) the underwater telephone, which furnishes voice communication between underwater craft and other ships.

Model JP Listening Equipment

DESCRIPTION

Models JP-1, JP-2, and JP-3 equipments are used on submerged submarines to obtain bearings on other vessels by directional detection of underwater sounds. They can be used also to listen for own ship's noise. Models JP-2 and JP-3 differ from JP-1 in the amplifier circuits. Models JP-2 and JP-3 are alike except for the method of mounting the hydrophone.

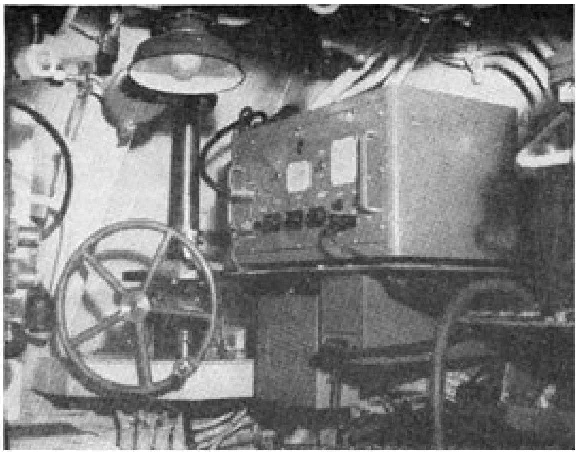


Figure 13-1 -Model JP-3 listening equipment.

In figure 13-1 the JP-3 receiver and training handwheel are shown mounted in a forward torpedo room. The hydrophone, which is not shown in figure 13-1, is mounted topside on a shaft operated by the handwheel. The training is manual.

The block diagram of the JP is shown in figure 13-2. With the aid of the tuning-eye indicator on the amplifier, the operator can train on a noise source with an accuracy of ±1 ½°. Relative bearings are read from a dial at the handwheel, as shown in figure 13-1.

Hydrophone

The hydrophone is not retractable. It is a directional line-type hydrophone 3 feet long. It is magnetostrictive and is polarized by permanent magnetization. Its frequency response is from 100 to 40,000 cycles per second. Because the hydrophone is mounted topside, the JP is sometimes referred to as "topside" listening gear.

Receiver

Figure 13-3 shows the circuit of the JP-1 audio amplifier with a line filter. The amplifier consists of four voltage amplifier stages and a power amplifier stage. The response is flat from 200 to 15,000 cycles per second. The amplifier response is still good above 20,000 cps, but the limit of audibility is about 15,000 cps.

The filter between the second and third amplifier stages is an *RC* filter that attenuates either high frequencies or low frequencies in five combinations. The filter switch, 5104, is a multiple constant selector switch having four sections. This switch has five positions, marked "*bass boost*," "*flat,*" "*500~*" "*3,000~*" and "*6,000~*" The *bass boost* filter reduces high frequencies to give a preference in response to frequencies near 150 cps. The *flat* filter gives a response that is essentially flat from 200 to 15,000 cps. The *500-cycle* filter attenuates low frequencies and passes high

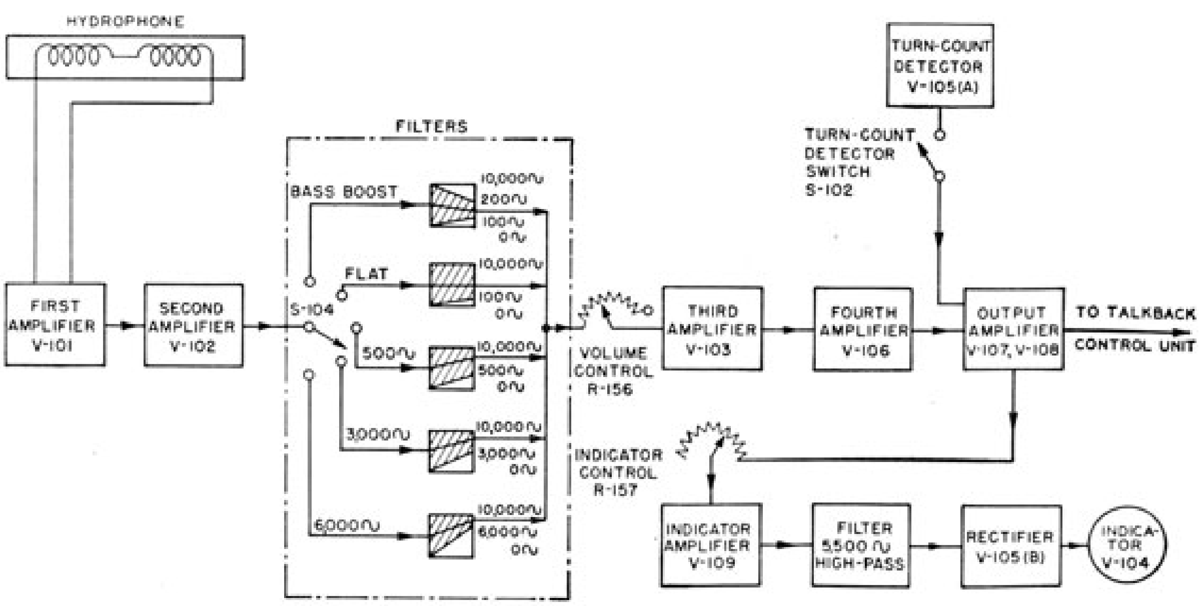


Figure 13-2 -Block diagram of the JP listening equipment.

FOLDOUT - Figure 13-3 -Circuit of the JP-1 audio amplifier with line filter.

frequencies above 3,000 cps without much attenuation. The *3,000-cycle* filter attenuates frequencies below 3,000 cps more sharply than the *500-cycle* filter. The *6,000-cycle* filter passes only a narrow band of frequencies in the vicinity of 6,000 cps. The *flat* alter is used normally when the operator is searching for a noise source. After a noise signal is received, one of the filters is selected to pass most of the noise signal and reject most of the unwanted background noises that are always present. The choice of filter depends on the frequency components of the signal.

The output amplifiers are two 6G6G tubes in push-pull. The output can be connected to loudspeakers, to headphones, or to an intercommunication "talkback" unit for relaying the output to the conning tower.

The output is connected also to the indicator-amplifier stage, V109, which further amplifies the signal. The signal then is passed through a high-pass filter, which greatly attenuates all frequencies below 6,000 cps. The high-frequency output of the filter is rectified and applied to the grid of the tuning-eye indicator. Only high frequencies are used because the directivity of the hydrophone is not adequate for low-frequency signals.

The stage marked "turn-count detector" in figure 13-2 is simply a diode that clips off the peaks of the signal input to the output amplifiers. The resulting distortion sometimes causes periodic noises, like propeller sounds, to stand out distinctly from water noises so that the operator can count the number of propeller beats per minute.

Model JT Listening Equipment

COMPONENTS

The model JT is a directional listening system designed to detect, identify, and locate sources of both sonic and ultrasonic sounds. It is designed to use the JP sonic equipment and has a *super-sonic converter* so that ultrasonic as well as sonic sounds can be amplified by the JP amplifier. In addition, it has a more directional hydrophone than the JP hydrophone and has a right-left indicator (RLI) for taking bearings on sonic sounds with greater accuracy than is possible with the tuning-eye indicator of the JP equipment. An interphone-amplifier unit permits "talkback" between the forward torpedo room-in which the JT system is mounted-and the conning tower.

Figure 13-4 shows the JT system. In this figure the supersonic converter, which permits the JP amplifier to be used with ultrasonic sounds, is mounted above the JP-1 amplifier.

The JT equipment uses a 5-foot line a 5F synchro receiver. The RLI is a pointer below the bearing indicator.

Block Diagram

A block diagram of the JT equipment, including the signal circuits, is shown in figure 13-6. The signal from the hydrophone can be connected either to the master control unit, or to the JP circuits shown above the control unit. The master control unit contains RLI circuits for taking accurate bearings on sonic sounds. The JP amplifier can be used with or without the supersonic converter.

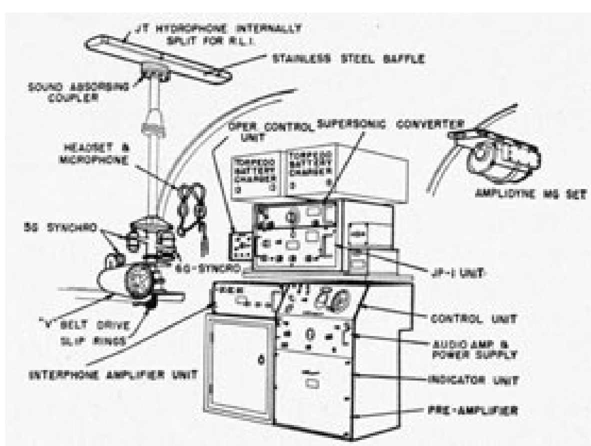


Figure 13-4 -JT listening system.

hydrophone. Because the JT hydrophone is longer than the JP, the JT has greater directivity. The bearing of the JT hydrophone is relayed by synchros to the control unit. The RLI is also on the control unit. The torpedo battery chargers shown in figure 13-4 are not a part of the JT equipment.

The master control unit (figure 13-5), shown below the JP-1 unit (figure 13-4), contains pre-amplifier, amplifier, and RM circuits. The bearing indicator is merely a bearing card attached to

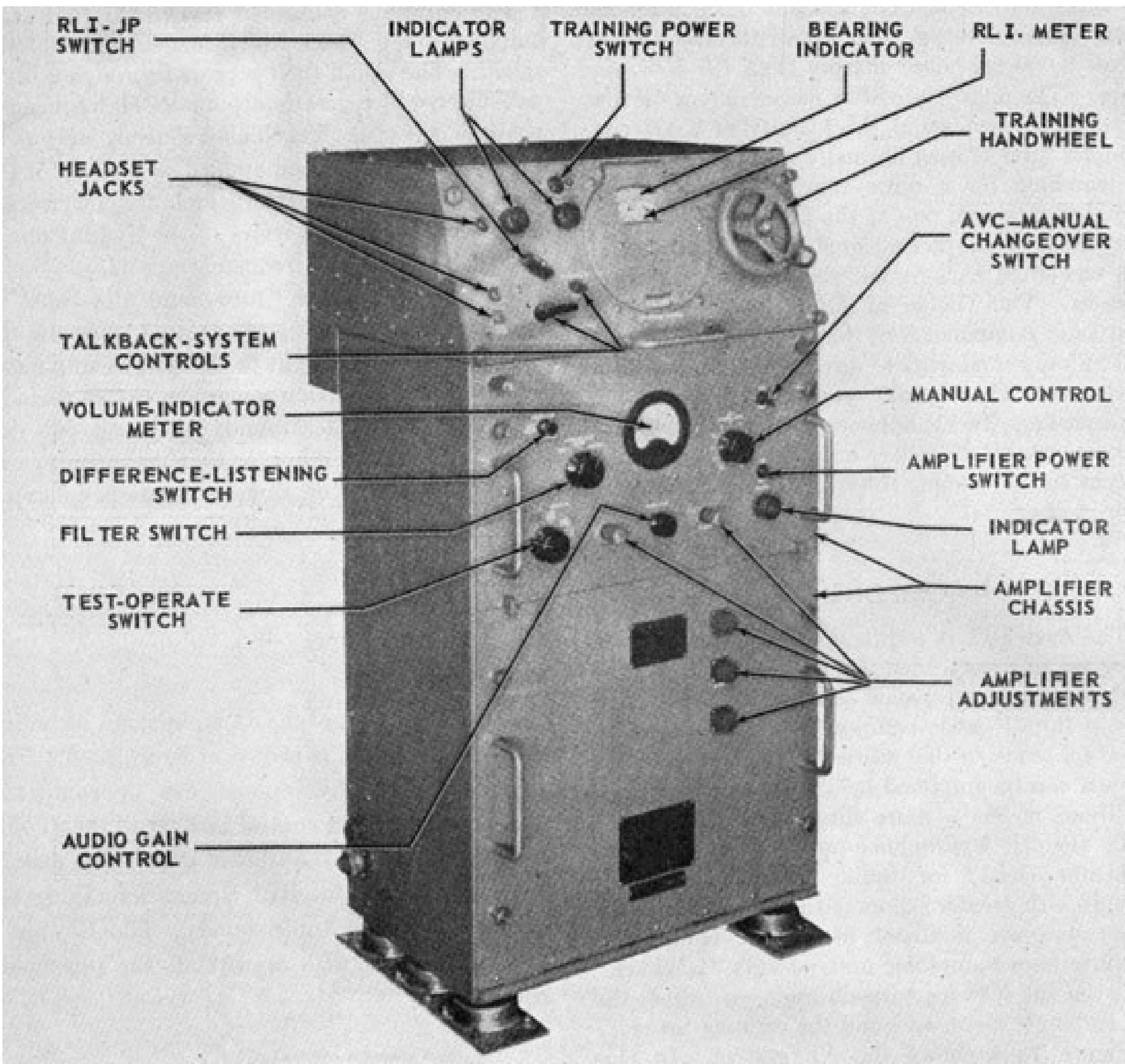


Figure 13-5 -Master control unit of the JT equipment.

The right-left indicator operates on the same principles as the bearing-deviation indicator (BDI) -that is, the two signals from the halves of the hydrophone are added, subtracted, shifted in phase, and then compared to indicate whether the hydrophone is trained to the right or left of the *on-target* position. The RLI makes it possible to take bearings on sonic sources to an accuracy of ± 1°.

The supersonic converter is used with only the JP amplifier for receiving ultrasonic noises up to 65,000 cps. The hydrophone signal is switched manually to the converter, which has oscillators and filters. The oscillators heterodyne the ultrasonic signals to sonic frequencies. The signals are then amplified in a part of the JP amplifier and are applied to the tuning-eye indicator for taking bearings on the source of the ultrasonic sound.

The JP amplifier can be used to receive either the output of the supersonic converter or the hydrophone output directly. It is seldom used in the latter way because when so operated it indicates the bearing of sonic sounds, and the RLI circuits in the master control unit indicate the bearing of sonic sounds with greater accuracy. Therefore, the RLI is used generally for sonic listening, and the supersonic converter and JP are used for ultrasonic listening. Although the RLI circuit has inherently greater bearing accuracy than the tuning-eye circuit of the JP, the latter has as good accuracy for ultrasonic listening as the RLI has for sonic listening because the hydrophone is more highly directional for signals of high frequency.

Training

The hydrophone is trained by a servo system operated from the master control unit. The hand-wheel on the master control unit is connected to the 5CT synchro, which controls the servo amplifier. The servo amplifier controls the amplidyne-type motor-dynamo amplifier, which operates the training motor. The bearing of the hydrophone is transmitted by synchro transmitter 5G to the synchro receivers in the conning tower and the master control unit.

A field-change kit has been supplied for the JT equipment. The kit adds *maintenance of true bearing* (MTB) and *generated target tracking* (GTT) to the training system. The MTB units compensate automatically for changes in the course of the submarine so that target tracking with MTB is smoother than unaided handwheel tracking. The GTT units provide the operator with aided tracking of a target designated at the fire control station. In GTT operation, computed target bearing from the fire control computer is checked against observed bearing to aid in tracking the incident sound to the length of the hydrophone, as shown in figure 13-7. At sound frequencies below 960 cps, the wavelength is longer than the 5-foot length of the hydrophone. When such a sound strikes the hydrophone, all the nickel tubes are subjected to equal pressure regardless of the orientation of the hydrophone, as shown in figure 13-7. However, when sounds of short wavelength strike the hydrophone, some tubes are compressed and others are expanded, depending on the orientation of the hydrophone. The coils are connected in series. Maximum response is obtained when the hydrophone is broadside to the incident wave, because in this case all of the voltages are series-aiding.

A steel and rubber baffle is mounted on the rear of the hydrophone to absorb sound coming from the rear. This baffle reduces the response of the hydrophone to sounds from the rear and prevents ambiguity in bearing measurement.

RLI OPERATION

Sum and Difference Inputs

The hydrophone is split into two halves. When the RLI-JP switch on the master control unit is switched to the RLI position, the hydrophone halves are connected as shown in figure 13-8.

The impulses in each coil of the hydrophone add vectorially. If the currents in the coils are like those shown in figure 13-8, the output of T101 obviously depends on the sum of the currents in the two halves of the hydrophone, whereas the output of T102 depends on the difference in the currents in the two halves.

Three relative positions of the hydrophone and target are shown in figure 13-9, A. When the hydrophone is oriented so that the wavefront strikes

target.

Hydrophone

The hydrophone consists of 10 nickel cylinders placed collinearly. Each cylinder is surrounded by a coil, in which an impulse is developed magnetostrictively each time the tube is compressed or expanded by a pressure wave.

The directional characteristics of the hydrophone depend on the ratio of the wavelength of the it at an angle, signals will be established in both the sum and difference channels. These will be exactly 90° out of phase.

Figure 13-9, B, shows the hydrophone signals for the three orientations shown in figure 13-9, A. Figure 13-9, C, is the vector representation of the signals shown in figure 13-9, B. Figure 13-9, D, shows the vector sum and difference of the signals. Note that the sum and difference signals at the input to the RLI circuit are always 90° out of phase. Note also that on one side of the true bearing the difference leads the sum, and that on the

other side of the true bearing the sum leads the difference. In the RLI circuit the difference signal is advanced by 90° with respect to the sum so that it is either in phase or 180° out of phase with the sum signal, as shown in figure 13-9, E. The difference signal is advanced 90° with respect to the sum, by advancing it 135° and by advancing the sum signal 45°, as shown in figure 13-10. The same effect could have been obtained by advancing only the difference signal 90°.

Amplifier Circuit

Figure 13-10, shows that the sum and difference signals from the hydrophone are amplified in two preamplifier stages and then passed through a filter that removes all frequencies below 500 cps and above 14,000 cps. The signals then pass through the *test-operate* switch. This switch is a six-position switch, three positions of which provide different amounts of attenuation of the signals.

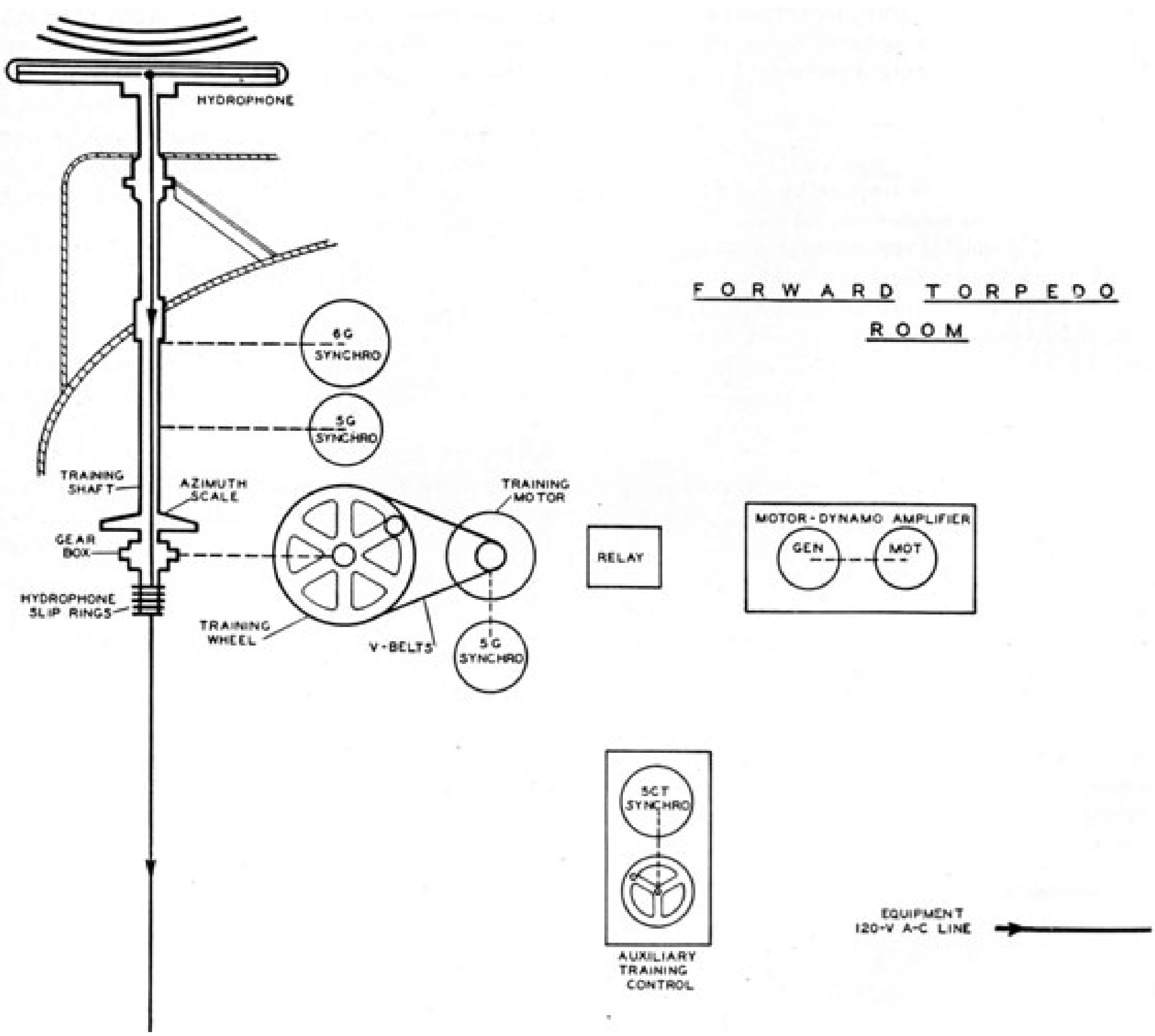


Figure 13-6. -Block diagram of the JT system.

The other three positions provide means for adjusting certain critical components.

After leaving the *test-operate* switch, the sum and difference signals are amplified further in two stages and then pass through a filter that attenuates all signals below 5 kc and above 9 kc. The signals then pass through the phase-shifting networks. The sum signal is advanced 45° in phase in its network, and the difference signal is advanced 135° in phase in its network. This phase shift makes the sum and difference signals either in phase or 180° out of phase, as explained previously.

After leaving the phase shifters, the sum and difference signals are applied to the first phase detector. Figure 13-11 shows the *first phase-detector* circuit and a Wheatstone bridge for comparison. This detector consists of a bridge circuit in which two of the arms contain series aiding diodes (the two sections of V208). The sum

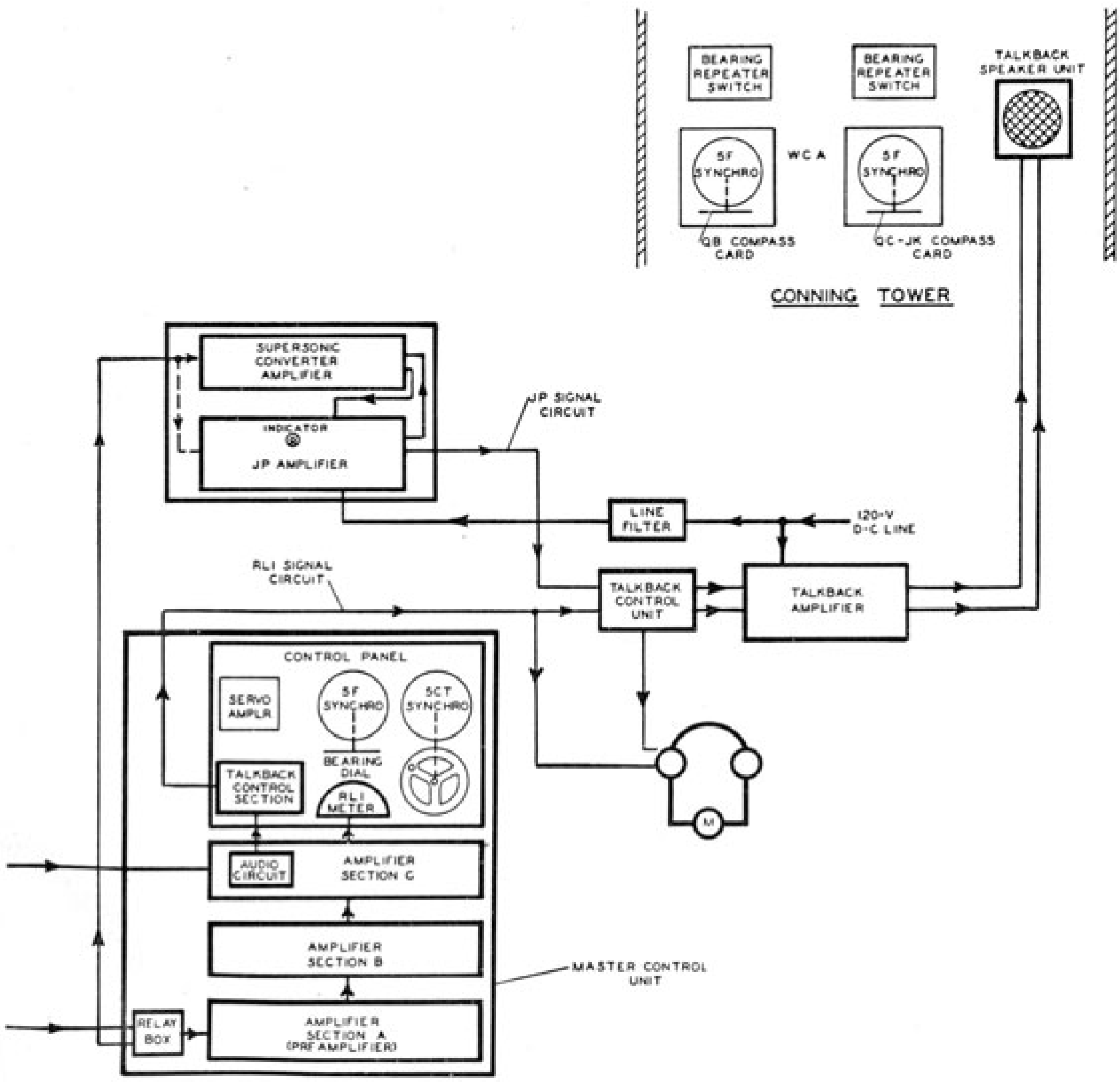


Figure 13-6 -Block diagram of the JT system-Continued.

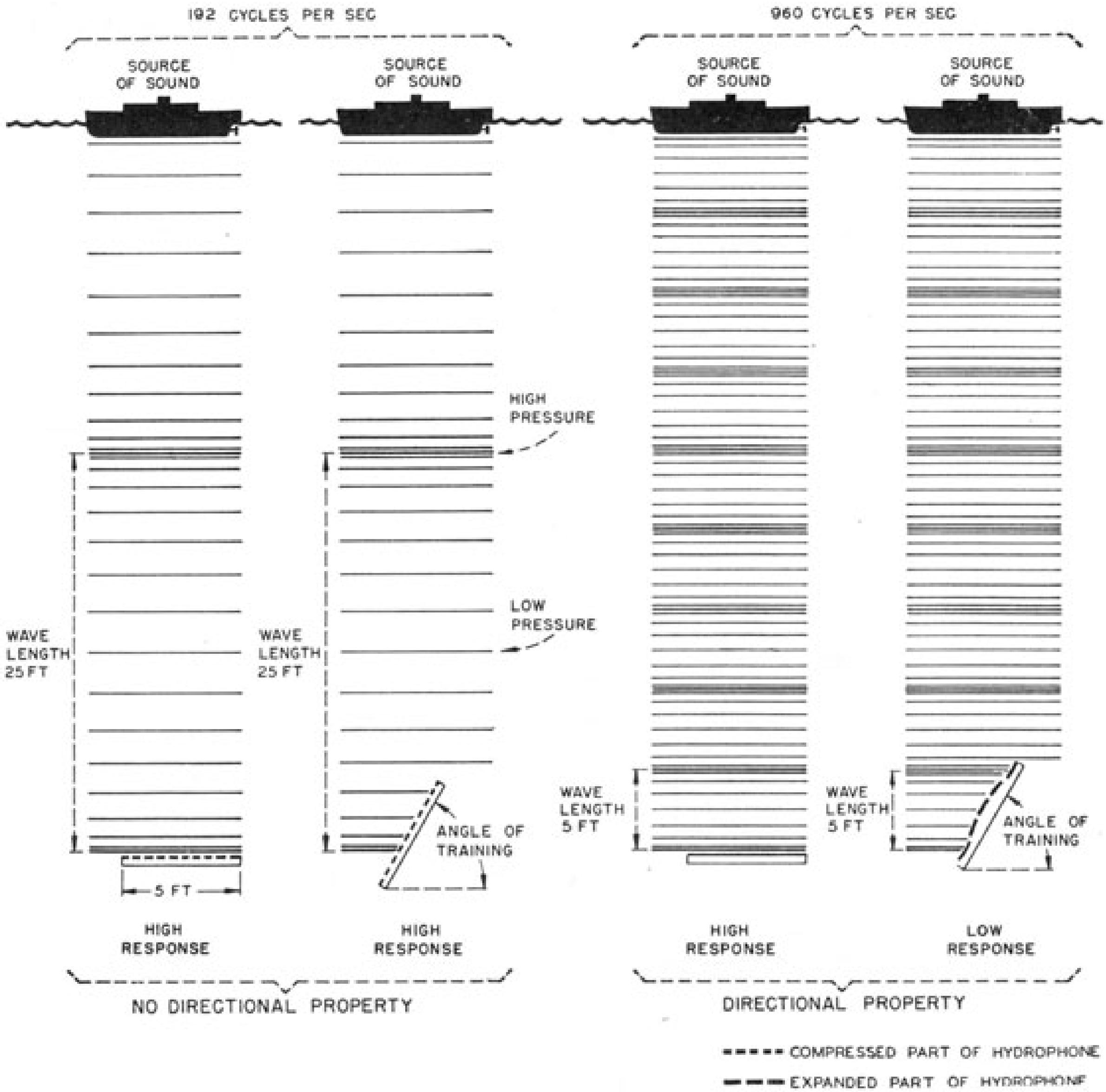


Figure 13-7 -Hydrophone response to sound waves of different wavelengths.

signal push-pull output from V207 is applied across opposite corners of the bridge and tends to make both diodes conduct simultaneously during one half of each cycle. The difference signal output from V206 is applied across R254 and R250 as a bias. The bias is applied simultaneously to the cathode of the first section of V208 and the plate of the second section thus biasing the two diodes with opposite polarity. Thus, one or the other of the diodes can conduct depending upon whether the difference signal is in phase or 180° out of phase with the sum signal. The output of the bridge is the d-c signal across R252. The polarity of this signal depends upon which diode conducts. The a-c component is bypassed to ground by capacitor C242 in shunt with the output.

The output of the bridge cannot be amplified and applied directly to the RLI meter because the amplitude distortion produced in the amplifier stages causes an erroneous indication of the meter. Therefore, the output of the bridge is interrupted by a 60-cps synchronous vibrator, CV-301,

amplified, and then is detected in the second phase detector, shown in figure 13-12. The *second phase detector* consists of a bridge circuit in which two of the arms contain series aiding diodes (the two sections of V304). A 60-cps reference voltage from transformer T302 is applied across opposite corners of the bridge and tends to make both diodes conduct simultaneously during one half of each cycle.

The output of the vibrator is a 60-cps square wave voltage which is amplified in V303 and appears across R328 and R329 of the bridge circuit as a bias. The output of V303 is applied simultaneously to the plate of the first section of twin diode V304 and the cathode of the second section hence biases the two diodes with opposite polarity. Thus, one or the other of the diodes can conduct depending upon whether the square wave signal is in phase or 180° out of phase with the reference voltage from transformer T302. The output of the bridge is a d-c signal across the RLI meter M501. The polarity of this signal depends on which diode conducts and determines in which direction the RLI meter will deflect.

The sum signal normally is connected to the audio amplifier for listening. However, the difference signal for listening can be selected by depressing switch 5301 (figure 13-10)-called the *press for difference listening* switch. The difference signal is selected when the operator has a large signal and desires to reduce the volume to "sharpen" his pattern. Reducing signal amplitude effectively sharpens the pattern because a small change in a small signal can be heard much more easily than the same change in a large signal.

Figure 13-13 shows the schematic diagram of the amplifier of the master control unit, with the RLI circuits and audio amplifier.

ULTRASONIC LISTENING

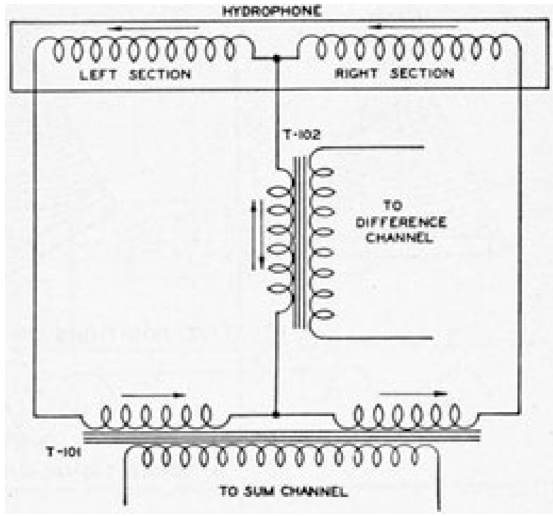


Figure 13-8 -Hydrophone connection for RLI operation.

is preceded by a filter that passes signals in a band of from 89 to 94 kc. Thus, the output is between 0 and 5 kc. Note that the output signal is zero frequency when the output of the first mixer is 94 kc.

The first heterodyning oscillator is adjustable within the frequency range of from 102 to 154 kc. The frequencies within this range can heterodyne with any signal of from 8 to 60 kc to produce an output difference frequency of 94 kc from the first mixer-corresponding to zero frequency audio output from the second mixer. Consequently, the first oscillator is calibrated in frequency within the range from 8 to 60 kc. The operator can measure the frequency of any incoming sound within this range by moving the calibrated dial until the beat frequency of audio output is zero. The frequency of the input signal is then indicated directly on the dial.

Because only signals between 8 and 60 kc are desired, the converter has a low-pass filter, actually a low bandpass filter, that attenuates all signals above 71 kc. In the first mixer all signals from 8 to 60 kc are heterodyned with the adjustable first oscillator frequency of 102 to 154 kc to produce an output difference frequency of 94 kc. The output of the first mixer is then heterodyned with the output of the 94-kc second oscillator to produce the audio-frequency signal. The output of the converter is connected to the third stage of the JP amplifier.

For listening to sounds of ultrasonic frequency, switch the hydrophone signal to the supersonic converter, as shown in figures 13-4 and 13-6. Figure 13-14 shows the block diagram of the supersonic converter, and figure 13-15 shows the schematic diagram of the converter.

The converter employs two mixers and two local oscillators. The first mixer raises the frequency and the second lowers it. The second oscillator is fixed at 94 kc and its associated mixer

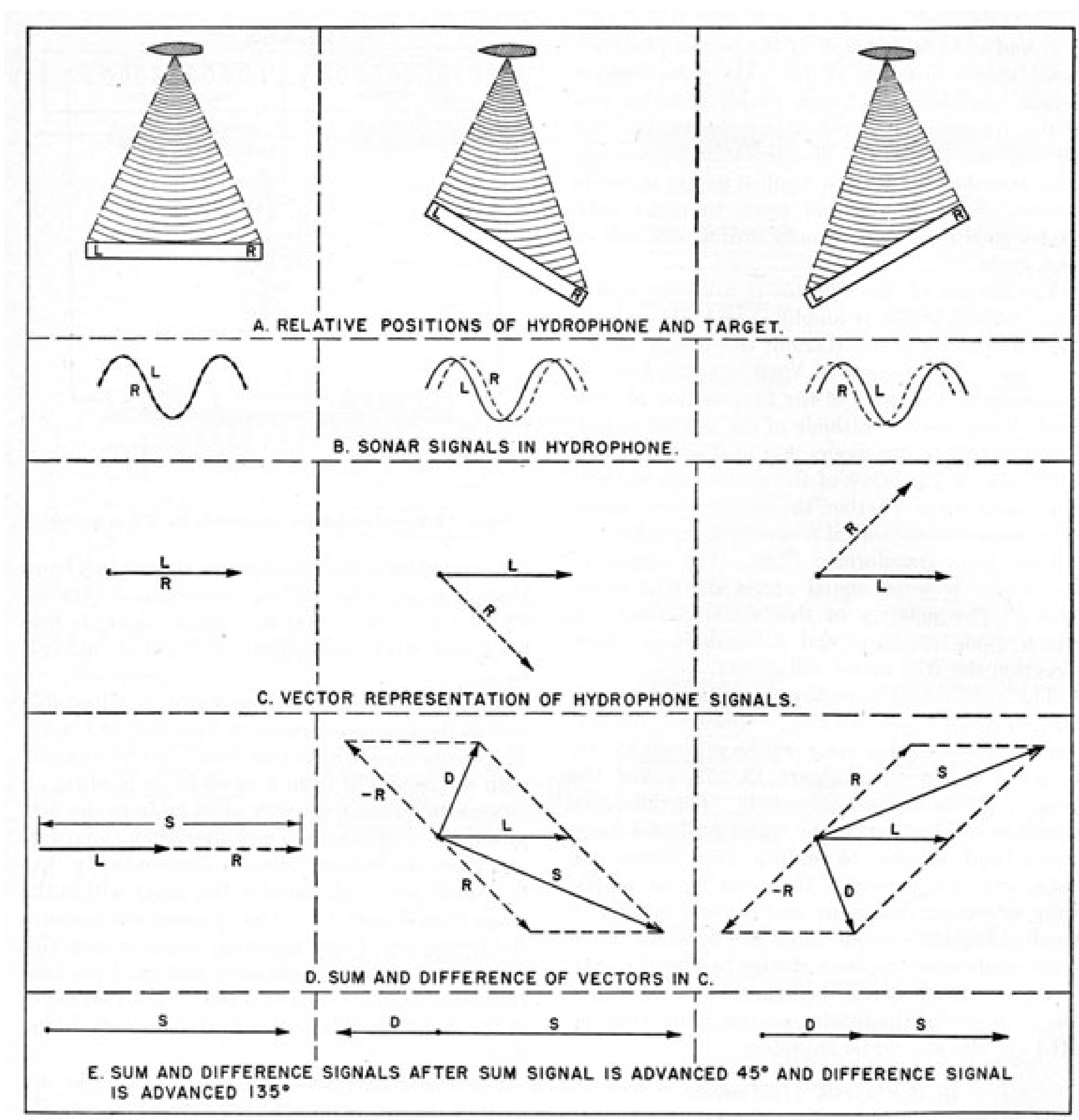


Figure 13-9 -Phase relations of signals in the sum and difference channels of the RLI circuit for various orientations of the hydrophone.

Triangulation-Listening-Ranging Equipment

GENERAL

In the first part of this chapter, methods of obtaining bearings with listening equipments have been discussed. Because the great advantage of a submarine over a surface ship lies in the fact that it can remain undetected until very late in the attack, or in some cases until after the completion of the attack, the use of listening equipment for determination of target bearings is of great importance to the submarine skipper. Targets can be located and accurate bearings taken at ranges up to 20,000 yards, without betraying the position of the attacking submarine.

In order to obtain ranges, however, the

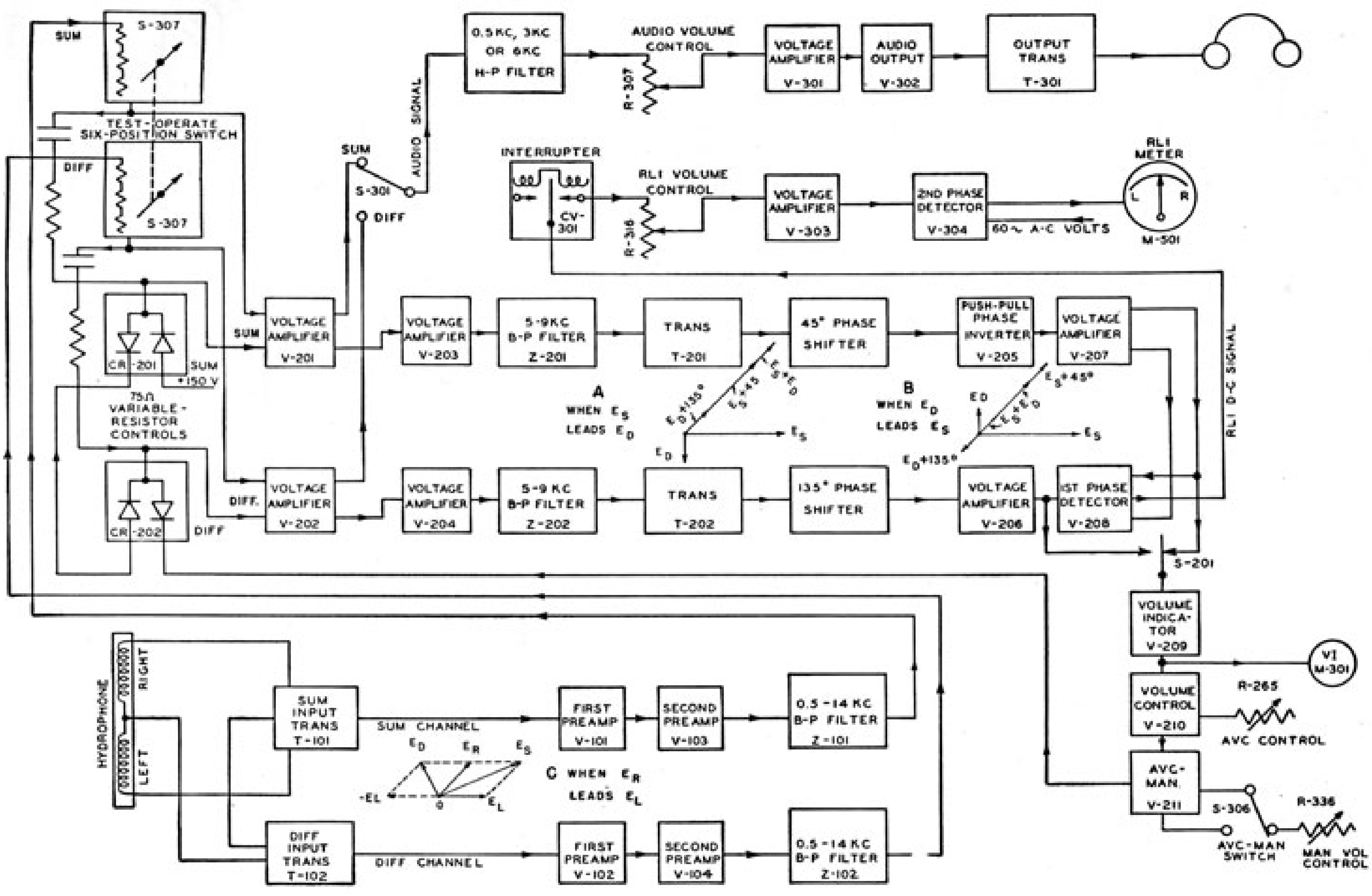


Figure 13-10. -Block diagram of the JT amplifier.

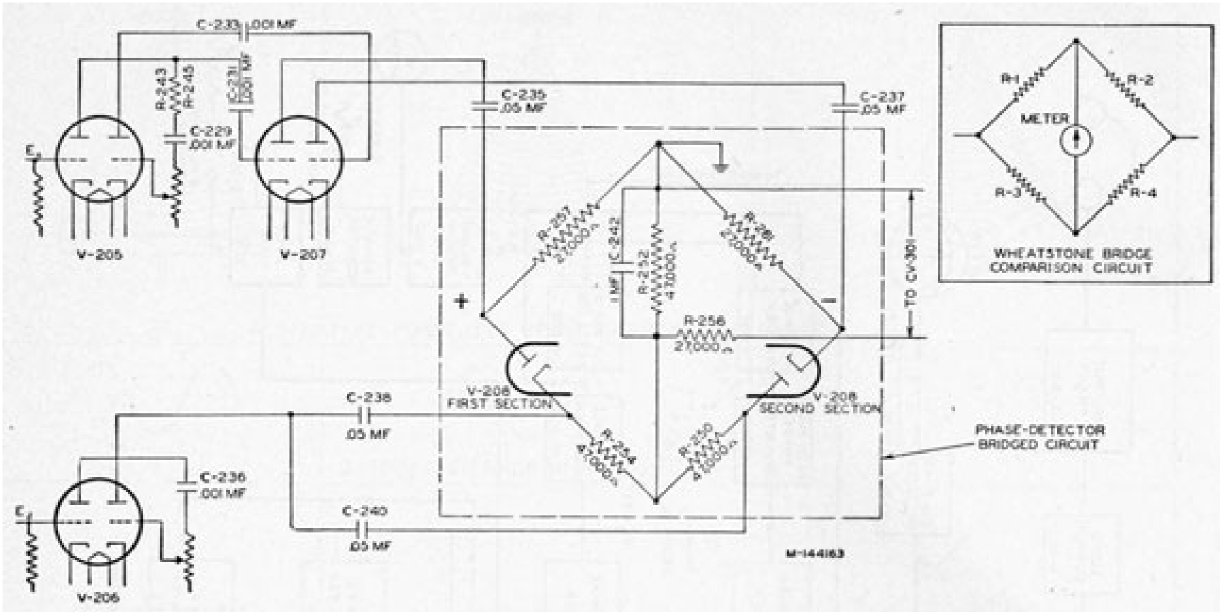


Figure 13-11. -First phase-detector circuit.

submarine must run the risk of detection by the enemy, and the possibility of losing a target. Until recently there have been three methods of ranging available to the submarine approach officer.

The oldest method is by optical means with the use of the range finder built as an integral part of the periscope. This method requires a knowledge of the target height, and is, at its best, inaccurate. In periods of low visibility this method is useless, and in periods of high visibility, the telltale wake of the periscope is easily detectable by the enemy lookouts.

The second method uses a radar whose antenna is built into the periscope. This method is extremely accurate, can obtain ranges at long distance s, and is reliable regardless of weather conditions, but the radar pulses can be received by enemy countermeasures equipment. The periscope must, of course, be exposed, and may be detected by the enemy ship either by visual methods, or by surface-search radar equipment.

The third method uses echo ranging. This method may be suitable when attacking targets which have no sonar equipment. However, most modern ships have facilities for reception of these ranging transmissions, which submarine, and allow the enemy to take evasive action and perhaps elude his attacker.

Thus it can be seen that the submariner has no effective means of ranging, with equipment discussed thus far, that will not reveal his presence. Extremely accurate bearings can be obtained with listening equipments by receiving supersonic frequencies which result in a very narrow reception pattern for the hydrophone. By placing one hydrophone near the bow of the submarine and another near the stern and using the length of the

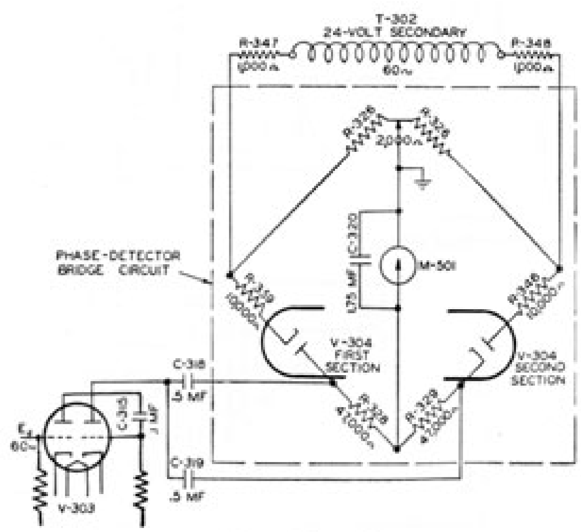


Figure 13-12. -Second phase-detector circuit.

would immediately indicate the presence of an attacking

FOLDOUT - Figure 13-13. -Schematic diagram of the amplifier of the master control unit.

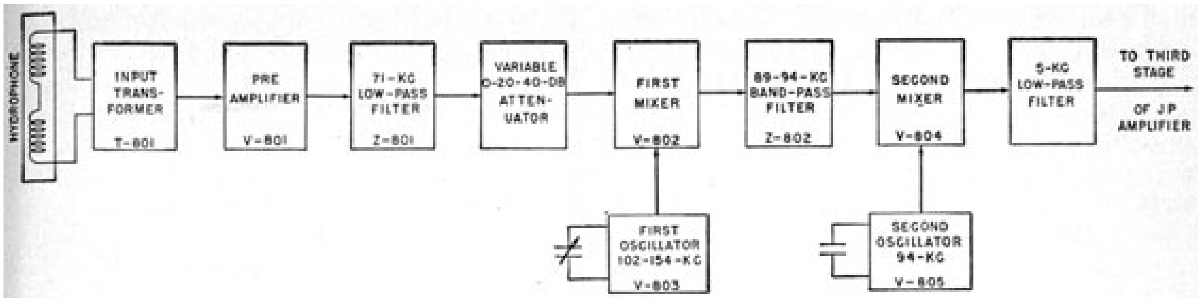


Figure 13-14. -Block diagram of the supersonic-converter unit.

submarine as a base line, the range could be determined by plane trigonometry. This arrangement would give the submarine commander a passive means of determining target range, without the disadvantages of the previously mentioned systems.

In figure 13-16 the general problem of ranging by triangulation is presented pictorially. The two hydrophones and the target form a triangle as shown. Angle *F* is the bearing angle of the forward hydrophone, *A* the bearing angle of the after hydrophone, *R* the range to the target from the forward hydrophone, *c* the distance from the after hydrophone to the target, *b* the distance between the forward hydrophone and the after hydrophone, and *C* the supplement of *F*-or 180°-*F*. From the law of plane trigonometry, known as the *law of sines*, the following relation is obtained:

$$R/\sin A = b/\sin B = c/\sin C$$

Angle *B* is equal to the difference of the forward hydrophone bearing angle and the after hydrophone bearing angle. Thus

$$R/\sin A = b/\sin (F-A)$$

$$R = (b \sin A) / \sin(F-A).$$

Angle *B* is always less than 15° in this application. The sine of a small angle is approximately equal to the magnitude of the angle in radians. Thus

$$R = (b \sin A) / (F-A). \quad (13\text{-}1)$$

Therefore, the range to a target is equal to the by the sine of the after hydrophone bearing angle and divided by the difference of the forward hydrophone bearing angle and the after hydrophone bearing angle.

DESCRIPTION

The following description of the operation of an actual triangulation-listening-ranging equipment, refers to the model JAA equipment. This equipment is shown in figure 13-17. Actually the JAA is an experimental model and will be replaced by another model for quantity production. however, the basic principles and modes of operation of the production model will probably be similar to those of the JAA.

Two methods of computing the ranges are used in the equipment. An electronic method, using an electronic range recorder, computes the range by receiving (1) a voltage proportional to the difference of the forward hydrophone bearing and the after hydrophone bearing by means of synchros and (2) a voltage proportional to the sine of the after hydrophone bearing, also by means of a synchro. The electronic range computer uses these voltages and the distance between the two hydrophones inserted as a constant to form a bridge. When the bridge is balanced according to equation (13-1), the range is the resultant, and is recorded on a chart.

A mechanical range computer computes the range in a like manner using gears, cams, and servo systems.

The range information and the forward hydrophone bearing information are sent to the torpedo data computer.

distance between the two hydrophones multiplied

239276°-53-17

Hydrophones

The hydrophones are identical to those used with the model JT equipment and are described

in an earlier part of this chapter. As in the JT equipment, the hydrophones are connected in halves for RLI operation. In addition to these hydrophones, two projectors called *squealers* are mounted on the submarine-one forward and one aft. They emit noises used for accurately aligning the hydrophone bearings with the baseline. These alignments must be made when contact with the enemy is not expected, as the noise emitted by these squealer projectors is easily detectable.

Control Stack

The control stack, with the mechanical range recorder, probably will be mounted in the conning tower. In the JAA equipment seven units, which provide the basic functions that would be necessary in any triangulation equipment, are mounted in the control stack. The units are (1) power supply, (2) forward bearing-deviation indicator, (3) after bearing-deviation indicator, (4) sonic a-f amplifier, (5) azimuth control, (6) servo electronic control amplifier, and (7) electronic range computer. Actually, in future equipments, some of these units may be installed in other locations and be operated by remote control in order to relieve congestion in the conning tower.

The power supply unit is of conventional design and supplies the necessary a-c and d-c voltages for operation of the various units.

The forward and after bearing deviation indicators are identical in operation and they closely resemble the BDI used in the JT sonar. The triangulation-listening-ranging equipment BDI's, however, provide a modulated a-c training control voltage for automatic target following in addition to BDI indication.

The sonic amplifier contains two identical channels which are used to amplify the signals for sonic listening. The forward channel amplifies the sum or difference signals from the forward BDI, and the after channel from the after BDI. Also incorporated in this unit is a noise generator which consists of a thermal oscillator followed by an amplifier, which produces a signal to energize either the forward or after squealer hydrophones.

The *azimuth control unit* contains the bearing repeaters, remote training controls, right-left meters, and a portion of the servo system used -for bearing repeating and range computation. The bearing repeaters consist of a forward repeater and an after repeater with a vernier dial that can be used selectively with either repeater, when the equipment is being operated manually. When in the automatic target-following mode of operation, this vernier dial indicates the difference between the forward and after hydrophone bearings. The RLI meters are conventional, and give an indication of whether the hydrophone is trained to the right or left of the target. This unit also supplies information to the electronic range computer for the range computation.

The *servo electronic control amplifier* is a three-channel control amplifier. The forward bearing servo channel controls the speed and direction of rotation of the forward bearing servo motor in accordance with the error voltages received from the azimuth control circuits. The difference angle servo channel is almost identical to the forward bearing servo channel. It controls the speed and direction of rotation of the difference angle servo motor, again utilizing the error voltage from the azimuth control unit.

Model OMA Noise-Level Monitor and Cavitation Indicator

GENERAL

Because noise is projected into the water by various equipments on the submarine, it is desirable to measure the noise level around the submarine at frequent intervals to assure that the noise level emanating from the submarine is not becoming excessive. The model OMA noise-level monitor (NLM) and cavitation indicator (CI) is designed to measure cavitation and other noises around own ship.

Cavitation is the formation of a vacuum around a propeller when the speed of the propeller exceeds a critical value. The vacuum is formed because the propeller pushes the water away from it at a

FOLDOUT - Figure 13-15 -Schematic diagram of the supersonic-converter unit.

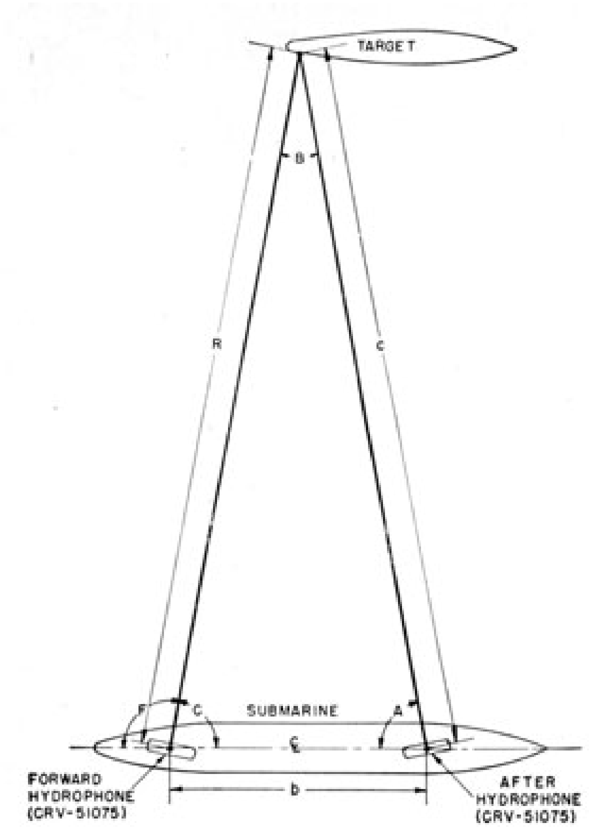


Figure 13-16 -Target ranging by the use of two hydrophones.

rate faster than the water can flow toward it. cavitation causes loss of efficiency and a high noise level. As cavitation is dangerous when the boat is maneuvering to avoid an enemy, an instantaneous indication of the beginning of cavitation is desirable.

DESCRIPTION

The model OMA equipment is shown in figure 13-18. It consists of an amplifier-indicator unit, a power supply, two neon-lamp cavitation indicators, and five hydrophones.

Four of these hydrophones are distributed along the pressure hull to detect noises at different locations. The fifth hydrophone, which is near the ship's screws, detects cavitation noise.

The equipment operates from a single-phase, 115-volt, 60-cps, a-c source. The schematic diagram of the amplifier of the OMA equipment is shown in figure 13-19.

The four NLM hydrophones and the one CI hydrophone are connected to the input. Switch 5101 selects one of the four hydrophones for monitoring the noise level. Switch S102nA, which selects either NLM or CI operation, is a spring-return switch and is normally set in the CI position. The hydrophone signal is amplified in two amplifier stages and then is filtered. For CI operation, a band-pass filter that passes a band of frequencies of from 6 to 12 kc is used. When switch 5102 is depressed for NLM operation, a filter that passes frequencies of from 150 to 3,500 cps, and a 20-step 60-db attenuator are connected into the circuit in place of the CI filter and the CI volume control, R113. The signal from the CI volume control or NLN attenuator is amplified in two additional amplifier stages.

For CI operation the signal is applied to the power amplifier, V106, for driving three neon lamps. The neon lamps are connected so that number 1 lamp flickers intermittently when the voltage across the secondary of T301 is 9 volts or more. The number 1 and number 2 lamps light when the voltage becomes approximately 18 volts. All three lamps light when the voltage is 25 volts or more.

For NLM operation the signal from V104 is connected to the cathode follower, V105. The DB meter that reads the noise level in the cathode circuit of V105, operates as a vacuum-tube voltmeter.

Sonar Communication Set AN/UQC-1

DESCRIPTION

The AN/UQC-1 equipment is designed for use in submarines and surface ships to provide voice or c-w communication through the water. As shown in figure 13-20, the equipment consists of a transducer, a receiver-transmitter unit, and a set-control unit.

The transducer has an omnidirectional pattern in a horizontal plane. The transmitter applies 400 watts of single-sideband, amplitude-modulated

energy to the transducer. Under favorable conditions, this power permits communication at ranges beyond 12,000 yards.

The carrier frequency is 8.0875 kc. The audio bandwidth of the modulator is from 250 to 3,000 cps. Because only the upper sideband is transmitted, the transmitted bandwidth is from 8,338 to 11,088 cps.

Figure 13-21 shows the block diagram of the equipment. Solid lines in the figure indicate transmission circuits; dotted lines indicate reception circuits. The same transducer is used for transmission and reception.

For voice transmission the microphone output is amplified in the speech amplifier and then clipped to maintain a constant output level. The modulator heterodynes the audio signal with the 8.0875-kc oscillator signal and also removes the lower sideband and carrier frequencies. The upper sideband is amplified in the drive amplifier, which drives the power amplifier. The power amplifier drives the transducer.

For c-w operation the telegraph key controls the conduction of a keying tube, which passes a 712-cps signal from an oscillator to the speech amplifier. This signal is also clipped to maintain a constant output level. The c-w signal is simply a 712-cps tone on the 8.0875-kc carrier. For reception the transducer signal is first amplified by the receiver amplifier and then heterodyned in the demodulator with the 8.0875-kc signal. The demodulated wave contains the audio component which is them amplified in the driver amplifier before it is sent to the speaker or headphones.

CIRCUIT

Figure 13-22 shows the schematic diagram of the L203, and L206, and various filter capacitors and bleeder resistors. It supplies negative bias potentials to the type-810 power amplifier tubes and to the receiver-amplifier unit. The bias supply also supplies voltage for operating the microphone, for reducing hum, and for the speech amplifier.

The third supply is the plate and screen supply. It is a conventional full-wave rectifier supply. A voltage-regulating tube, V113, regulates a part of the output of the supply. The regulated output is used for the oscillators and clipper stages.

The fourth supply uses two type-3B28 rectifiers to develop the high voltage for the power-amplifier stage.

Carrier Oscillator

The 8.0875-kc carrier is obtained by dividing a 16.175-kc signal, which is generated by a crystal-controlled oscillator, V114. The 16.175-kc signal from the oscillator is limited by clipper-rectifiers CR103 and CR104 to aid in frequency stabilization. The clipped signal is fed to grid 1 of the pentagrid-mixer, V115.

The band-pass filter, Z102, in the plate circuit of the pentagrid mixer is a low-Q tank tuned to 8.088 kc. The signal at the plate of the converter is fed to grid 3 through C120. The feedback signal is predominantly 8 kc because the plate tank is tuned to 8 kc and the phase-shift in the feedback circuit cuts off the converter tube on every second cycle of the 16-kc input. Thus, the plate tank is shock-excited at one half the crystal-controlled frequency-that is, 8.0875 kc. The 8.0875-kc signal is amplified in V116 and sent to the modulator.

Audio Oscillator

AN/UQC-1 equipment. The relay-operated switch, 0301, is operated from the set-control unit.

Power Supplies

The AN/UQC-1 has four rectifier-type supplies, which are located in the power amplifier. The first is a full-wave bridge rectifier for supplying 120-volts d-c to all the relays used in circuit switching.

The second supply is the bias supply. It consists of rectifier tube V205, filter chokes L202,

Tube V106 is a phase-shift audio oscillator. The three-section feedback network between plate and grid produces the feedback necessary to establish 712-cps oscillation that is used in the speech amplifier when the telegraph key is depressed.

Receiver Amplifier

The receiver amplifier is a conventional two-stage RC coupled amplifier with cathode-follower output. The gain of the amplifier is varied by adjusting the bias on the variable-mu stages.

FOLDOUT - Figure 13-17 -JAA triangulation-listening-ranging equipment.

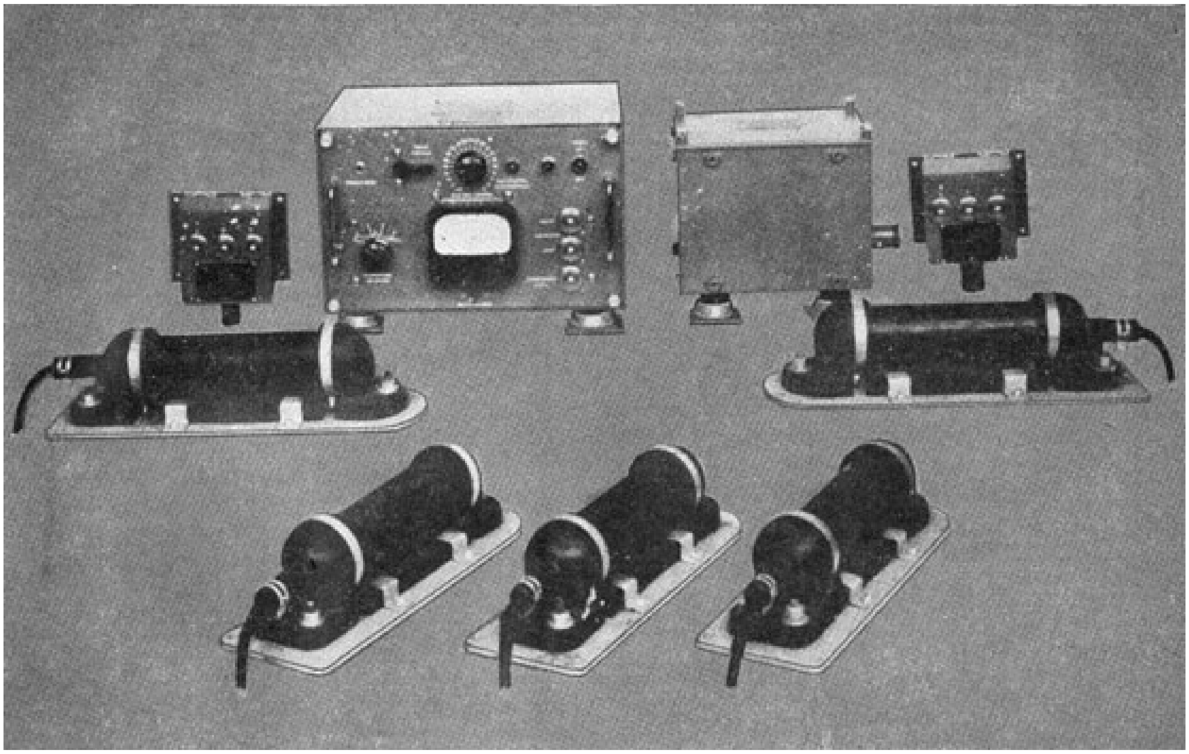


Figure 13-18.-Model OMA noise-level monitor and cavitation indicator.

Speech Amplifier

The speech amplifier consists of (1) an oscillator-lying network, (2) audio amplifier, and (3) limiter.

The oscillator-keying network uses tube V105. The 712-cps signal used for c-w transmission is applied to the grid of V1O5B at all times. When the telegraph key, K401, is depressed, V105B becomes conducting and the 712-cps signal is amplified and used in the speech amplifier. When the key is not depressed, V1O5B is cut off by the bias voltage supplied by voltage divider R131, R132, and R133 in conjunction with the reduced c-w) signal to the modulator. A cathode follower V103B, is used to send the signal input to the modulator.

Modulator

The modulator consists of T104, T105, two filters, and rectifier assembly CR105. It is a conventional balanced modulator. The audio modulating signal (c-w or voice) is applied across T105. The filter, Z1O1A, attenuates all modulating frequencies above 3 kc. The 8.0875-kc carrier is applied between the midpoints of transformers T104 and T105. The carrier and audio signals are heterodyned in the

plate voltage caused by V1O5A which has the same plate load resistor as V105B and a positive bias. The microphone output is connected to the grid of V104A. V1O4A and V104B constitute a two-stage feedback amplifier.

The limiter consists of two type-1N34 crystal rectifiers, which limit the modulating (voice or nonlinear circuit assembly CR105, which contains four type-1N40 crystal rectifiers. Because of the center tap connections in the balanced modulator, the carrier is cancelled and does not appear in the output of T104. Only sum and difference frequencies reach the secondary of T104. The band-pass filter,

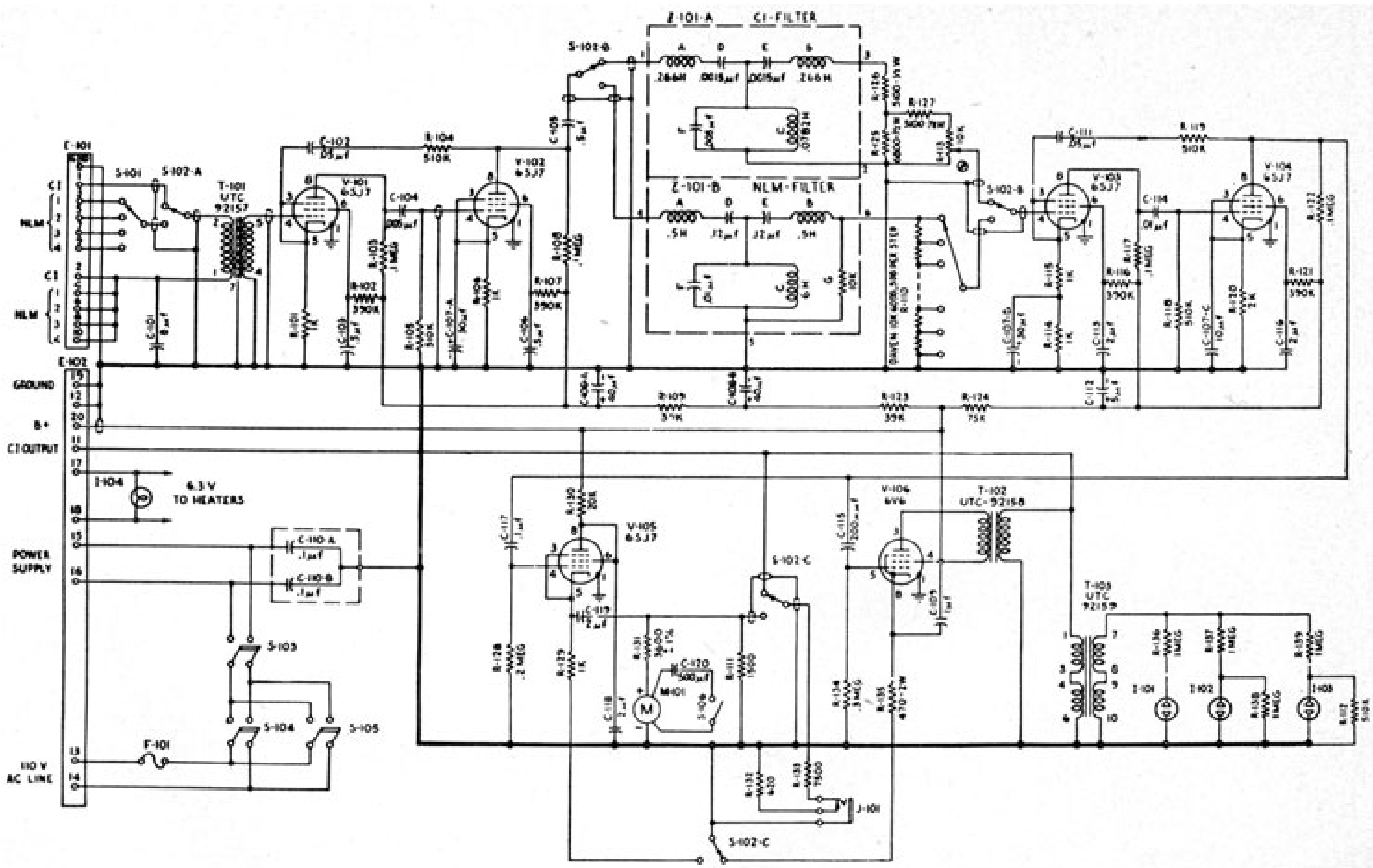


Figure 13-19. -Schematic diagram of the OMA amplifier.

Z101B, passes only the upper-sideband (sum) frequencies and attenuates the lower-sideband (difference) frequencies.

Driver Amplifier

The driver amplifier uses conventional push-pull output with negative feedback. It consists of amplifier V107, phase-inverter V108, push-pull driver stages V109 and V110, and push-pull output stages V111 and V112. The output is used to drive (1) the power amplifier when transmitting and (2) the loudspeaker when receiving.

Power Amplifier

The power amplifier comprises a pair of type-810 triodes in push-pull. When supplied with a 2,000-volt B+ supply, these tubes can deliver 400 watts of power to the transducer. The output transformer, T202, is provided with two taps to match the tubes to the load. The matching is done by connecting the transducer to the tap that results in maximum voltage across the output of T202.

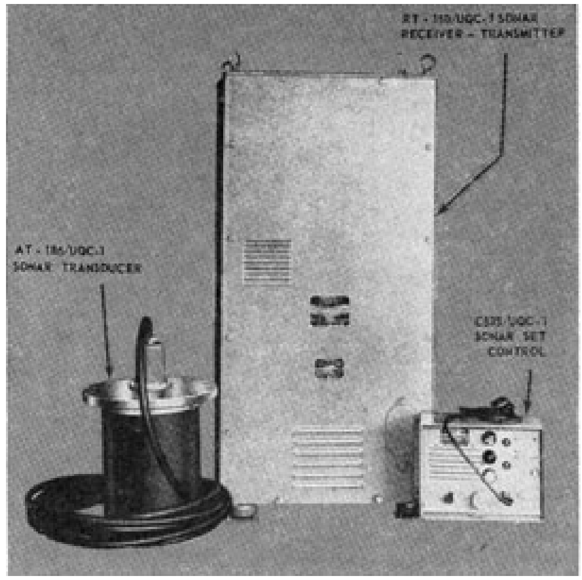


Figure 13-20. -Sonar set AN/UQC-1.

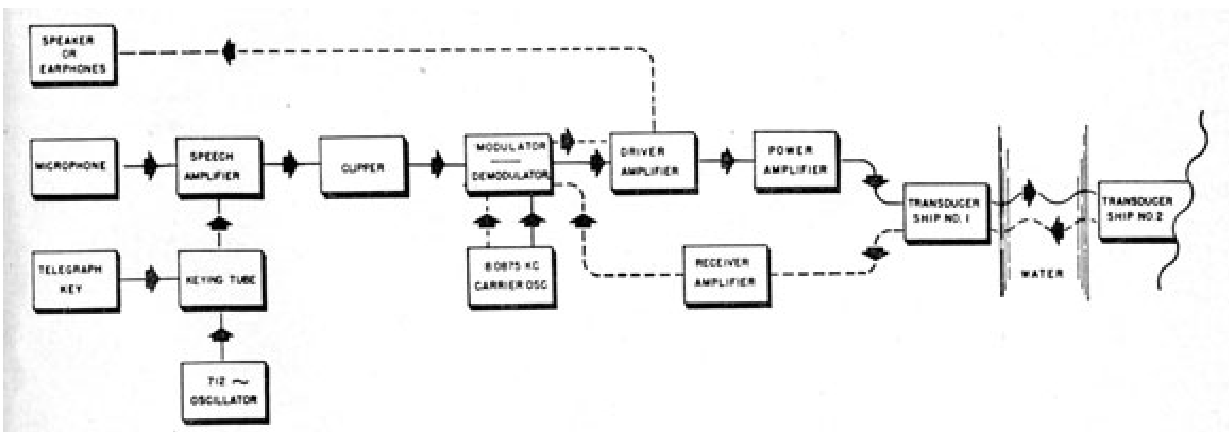


Figure 13-21. -Block diagram of the AN/UQC-1 equipment.

CHAPTER 14

SUBMARINE SONAR EQUIPMENT

Introduction

The primary method of sonar detection by a submarine is listening (chapter 13) because echo ranging discloses its presence. On a war patrol a submarine maintains a listening watch. When a target is detected, the bearing is determined by the listening equipment. When the sonar range to the target is desired, the echo-ranging transducer is trained to the target bearing and a single short ping is emitted. The echo-ranging equipment on a submarine is used most often for navigation and only as required for target ranging.

During World War II, combination sonar equipments that provided ranging, listening, and sounding were installed on submarines. The model WCA is such a combination sonar equipment.

The model WFA sonar equipment, which has torpedo-detection and mine-detection circuits, was developed near the end of World War II. As the WFA does not have a sounding device, it is installed usually with an echo-sounding equipment, such as the model NGA. The WCA and WFA-1 are described in this chapter. The NGA is a sounding equipment for installation in submarines only. It provides a paper record as well as a rotating-light indication.

Capacitive and f-m scanning sonars, which have been developed since the war, are being installed on modern submarines. The model QHB-1 capacitive-type scanning sonar and the model QLA f-m scanning sonar also are described in this chapter.

Model WCA Sonar Equipment

The WCA sonar equipment was mounted on most submarines during World War II. It is now being replaced by scanning equipment. Because it is no longer being installed on modern submarines, the WCA sonar equipment is described only briefly.

The model WCA-2 equipment uses 3 transducers and 1 hydrophone in three separate housings, as shown in figure 14-1. The NM is a magnetostriction transducer. The QC magnetostriction transducer and the JK Rochelle-salt hydrophone are housed in one sound head, called the QC-JK. The QB is a Rochelle-salt transducer.

As shown in figure 14-2, the WCA-2 consists of three systems.

One system, the QC-JK, uses the combination sound head for echo ranging and listening. The QC magnetostriction element is used for echo ranging. The JK crystal hydrophone, which is more sensitive than the QC magnetostriction transducer, is used for listening only. The QC and JK units cannot be used simultaneously.

The second system uses the QB Rochelle-salt transducer for echo ranging. The QB transducer can be operated over a wider range of frequencies than the QC because a crystal transducer is less sharply resonant than a magnetostriction transducer of the same beamwidth.

The third system uses the NM transducer for sounding only. The NM system requires no training controls because the beam is directed vertically downward.

Although the combined equipment consists essentially of three separate complete systems, each of the three systems uses one or more units in common with one or both of the other systems.

FOLDOUT - Figure 13-22. -Schematic diagram of the AN/UQC-1 equipment.

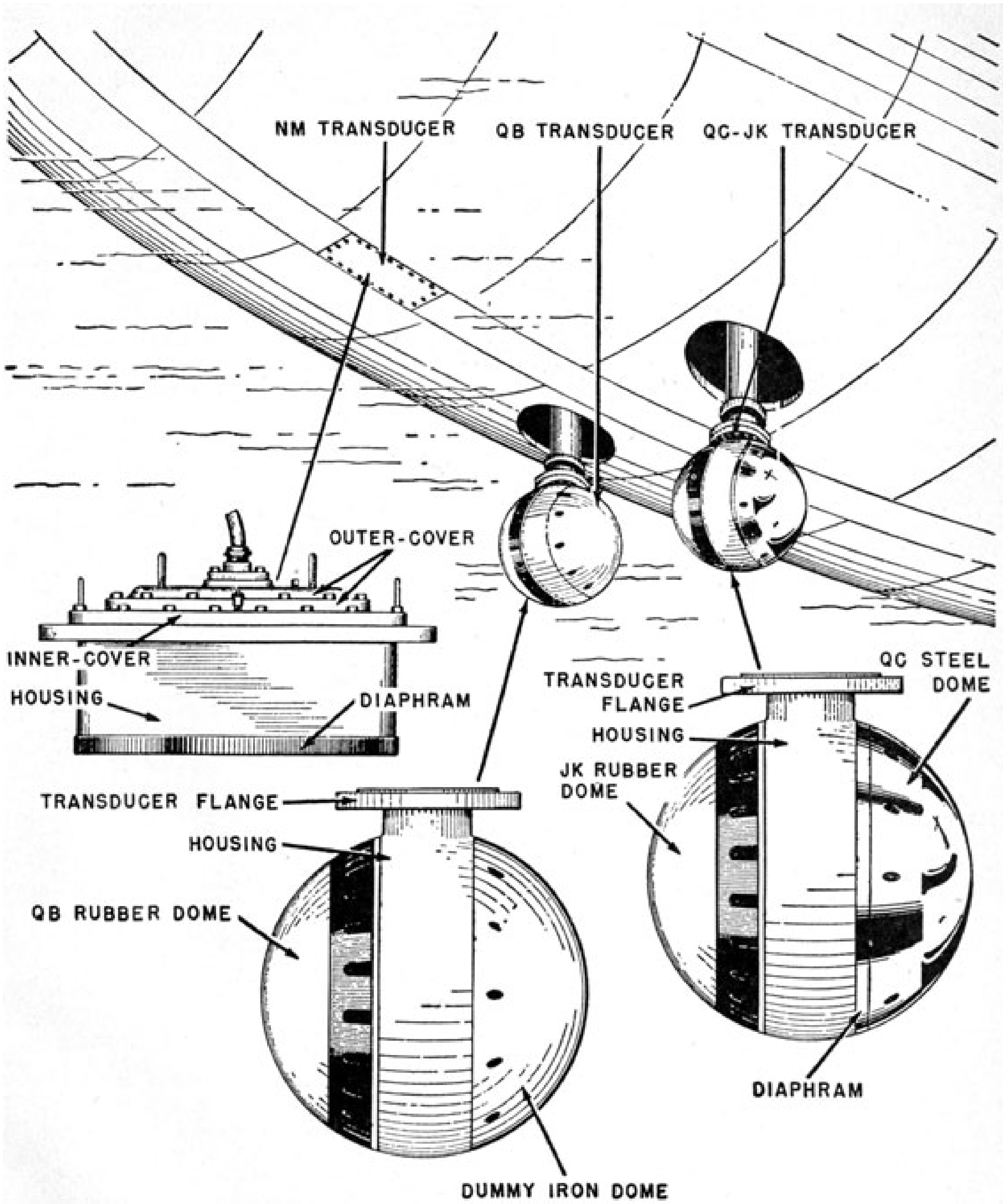


Figure 14-1. -The three sound heads of the WCA-2 sonar equipment.

261

For example, the QB and QC-JK systems use the same range indicator, and the sounding and QC-JK systems use a common driver unit.

The units of the WCA are similar to standard units discussed previously. The indicator is the familiar rotating-light type used on early echo-ranging equipment. The sounding unit has its own indicator, which is a rotating-light type. The only major difference between surface-ship echo-ranging equipment and the WCA is the use in the WCA of a Rochelle-salt echo-ranging system. The remote training unit provides for slewing in either direction because the equipment is used for listening most of the time.

Model WFA-1 Sonar Equipment

The model WFA-1 is a searchlight type of echo-ranging and listening equipment. It can be operated in any one of the following modes: (1) Listening, (2) echo-ranging, (3) communication, (4) torpedo-detection, (5) mine-detection, and (6) monitoring own ship's noise.

Listening may be carried out over the frequency range of from about 200 cps to about 100 kc. Bearing is determined accurately by a bearing-deviation indicating (BDI) meter.

The frequency band for echo ranging is from 17.2 to 46 kc. The BDI circuits can be used with echo-ranging operation for high accuracy. Higher frequencies used in enemy waters give maximum secrecy and highest bearing accuracy, but shorter maximum ranges.

Telegraphic communication with other vessels equipped with echo-ranging equipment is made possible by the inclusion of a telegraph key that keys the transmitter. The frequency range

The WFA-1 equipment has two identical control stacks, one in the conning tower and the other in the forward torpedo room. The conning tower is the primary control point, but, in an emergency, control can be switched to the forward torpedo room. Figure 14-3 is an over-all pictorial diagram of the equipment.

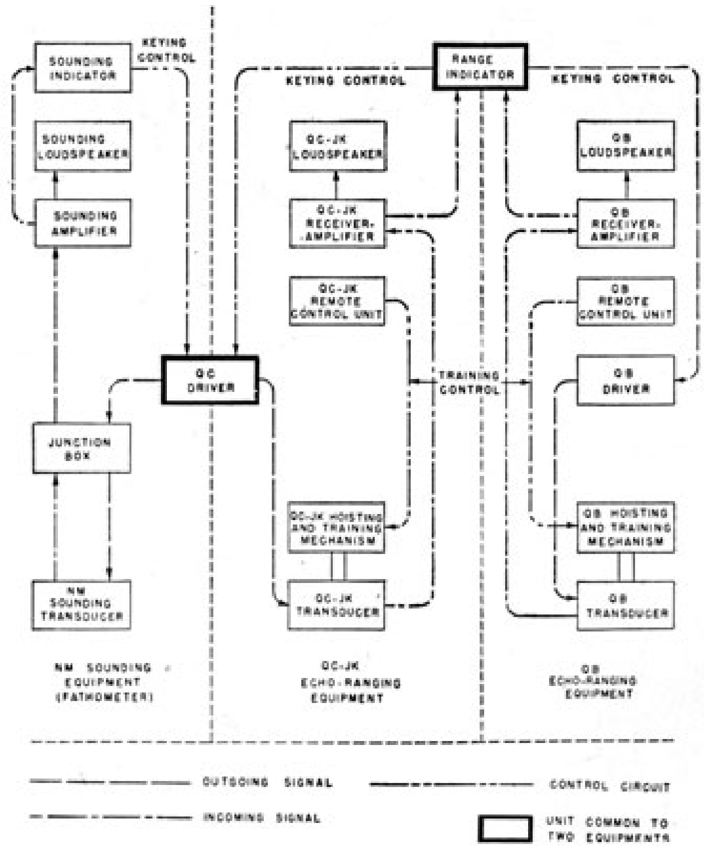


Figure 14-2. -Block diagram of the WCA-2 sonar equipment.

The WFA-1 has two sound head assemblies-one mounted on the deck, the other on the hull near the keel.

The topside sound head is mounted vertically on the main deck above the forward torpedo room

for this mode of operation is the same as for echo ranging.

For torpedo detection, the transducer beneath the keel is rotated constantly. Any sound signal picked up is fed through the receiver to the range recorder, the stylus of which is synchronized with the transducer rotation, as explained later.

For mine detection, the equipment is operated as a short-pulse echo-ranging equipment using either the topside transducer or the transducer beneath the keel. In this mode of operation the transducer is trained over a restricted arc of 30° or 40° on each side of the bow. Mines and other small navigational hazards can be detected at ranges up to 600 yards. The short pulse produces the high resolution that is necessary to detect objects, such as mines at short ranges or in close proximity to one another.

and is rotated by the topside training mechanism. It is not retractable. This sound head consists of three individual units, as follows:

1. A low-frequency (sonic) hydrophone, which operates over a band of from 200 cps to 15 kc for listening.

2. An intermediate-frequency (ultrasonic) transducer, which operates over a band of from 17.2 to 35 kc for echo ranging, and over a band of from 12.5 to 35 kc for listening.

3. A high-frequency (ultrasonic) transducer, which operates over a band of from 35 to 46 kc for echo ranging and over a band of from 31 to 100 kc for listening. These frequencies are selected because the crystals that are used operate at optimum efficiency in these ranges.

The lower sound head is mounted on a hoist-train shaft and functions like any retractable searchlight transducer. It contains a single crystal transducer that operates over a band of from 22 to 32 kc, with most efficient operation at 27 kc for both echo ranging and listening. The sound head is spherical to prevent turbulence at high speeds and to keep water noise at a minimum.

For torpedo detection, the lower sound head is rotated at 12 revolutions per minute and the range-recorder stylus sweeps across the chart in about 4.6 seconds. Thus, each sweep of the stylus occurs in one complete revolution of the transducer. Stylus travel is synchronized with the transducer by a microswitch mounted on the transducer shaft. The interval for fly-back of the recorder stylus occurs while the transducer rotates through the sector from 170° to 190°. This sector is chosen for fly-back because during this interval the transducer sweeps across the ship's stern, and only unwanted sounds from the screws are picked up. The bottom scale on the recorder is graduated to indicate the bearing of all received sounds. Torpedo detection is strictly a listening function.

The receivers are of the sum-and-difference type and have BDI meters, as well as RCG circuits.

Model QHB-1 Capacitive-Scanning Sonar Equipment

The model QHB-1 is almost identical with the model QHB scanning sonar that is used on surface ships. The QHB-a is described in chapter 6 of this text. The QHB-1 differs from the QHB-a in the following respects: (1) the QHB-1 has a relative instead of a true PPI presentation-that is, the bearings of signals are referred only to the heading of the submarine; (2) it has a single-ping keying feature as well as facilities for automatic keying; and (3) its transducer is more ruggedly constructed to withstand increased pressure without leaking. To prevent accidental keying, the keying control mechanism of the QHB-1 must be manually held in the closed position, either for "automatic" or repetition keying or for single-ping operation.

Model QLA F-M Scanning Sonar Equipment

DESCRIPTION

The QLA echo-ranging equipment is an f-m scanning sonar. It provides a plan-position indication (PPI) of underwater objects within sound range. It can be installed on submarines or surface vessels. In contrast to the searchlight-type sonar, the f-m sonar provides continuous area search coupled with the ability to detect very small objects.

Sonar echo-returns from vessels, wakes, sand banks, antisubmarine nets, and other submerged objects that reflect ultrasonic energy are presented both audibly and visually. The audible signals are tones. The visual signals are intensity-modulated spots on the oscilloscope PPI indicator. Figure 14-4, A, shows a surface ship entering a channel. Figure 14-4, B, shows the QLA indication aboard the ship.

The QLA equipment and the location of the major units are shown pictorially in figure 14-5. The major units are (1) the *frequency-modulated oscillator*, which generates the f-m signal; (2) the *driver*, which amplifies the f-m signal and drives

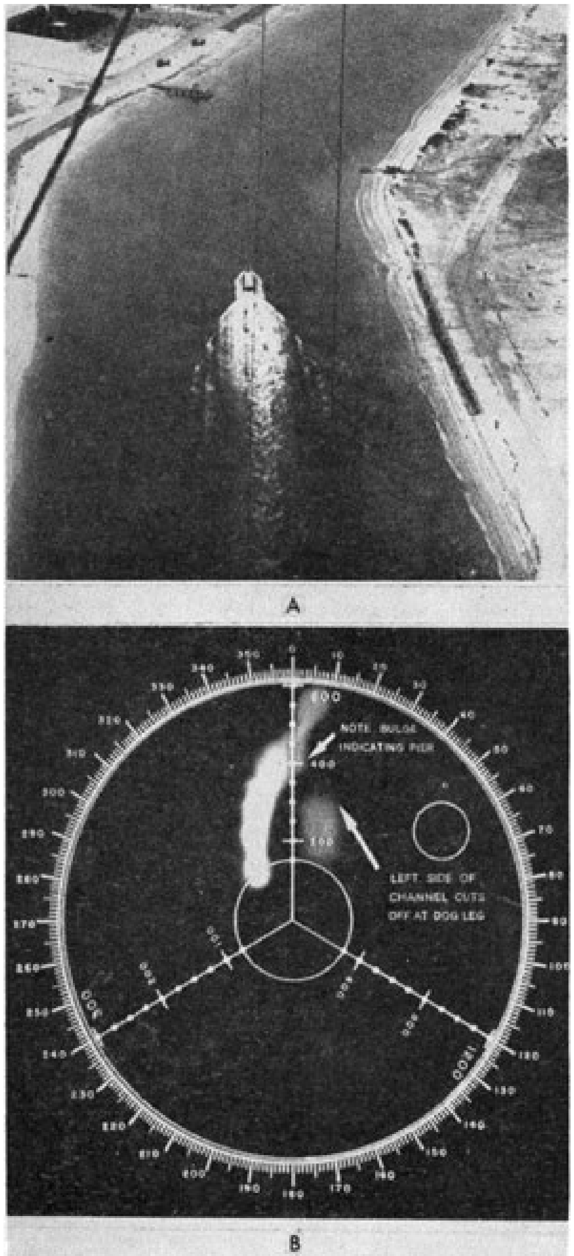


Figure 14-4. -Typical QLA indication. A, Ship entering channel; B, indication aboard the ship.

the projector; (3) the *sound head*, which contains the projector and a hydrophone for receiving the echo returns; (4) the *hoist-train mechanism*, which raises, lowers, and rotates the sound head; (5) the *receiver*, which heterodynes the returning echoes with the oscillator signal; (6) the *analyzer*, which uses a series of 20 filters which are sequentially connected through electronic switch tubes to extract the frequency components of the heterodyned signal; and (7) the *indicator*, which intensity-modulates a cathode-ray tube beam in accordance with the output of the analyzer and which moves the beam to present a plan-position indication. The loudspeaker makes the returning echo audible, and the test oscillator is used to adjust the sweep of the frequency-modulated oscillator.

PRINCIPLES OF OPERATION

The functional block diagram of the QLA equipment is shown in figure 14-6. The f-m oscillator develops the carrier signal, which varies with time. At the beginning of an operating cycle the frequency is at a maximum of 46 2/3 kc, and at the end of the cycle (several seconds later) it is at a minimum of 36 kc. The frequency decreases uniformly from the maximum to the minimum value. At the end of an operating cycle the frequency returns abruptly to its maximum value and the cycle is repeated. The abrupt return or fly-back to initial frequency requires only a few milliseconds, during which time the projector is silenced or "blanked."

If frequency is plotted vertically and time horizontally, the result is a curve having a sawtooth pattern, as shown in figure 5-20. The downward slope of the sawtooth signal represents a decrease of frequency with time; the vertical line forming the left side of the waveform represents the fly-back from minimum to maximum frequency. The length or base of the sawtooth waveform corresponds to the time required for one operating cycle. The frequency at any instant is different from that at any other instant in an operating cycle, and the frequency changes linearly with time.

The QLA sound head contains a projector and a hydrophone. The projector transmits sound waves in a wide fan-shaped beam that has an arc of about 80°. The sound waves are reflected by any object in the beam of the projector. A small part of the reflected energy returns to the sound head as an echo, where it is picked up by the sharply directive hydrophone.

FOLDOUT - Figure 14-3. -Over-all pictorial diagram of the WFA-1 equipment.

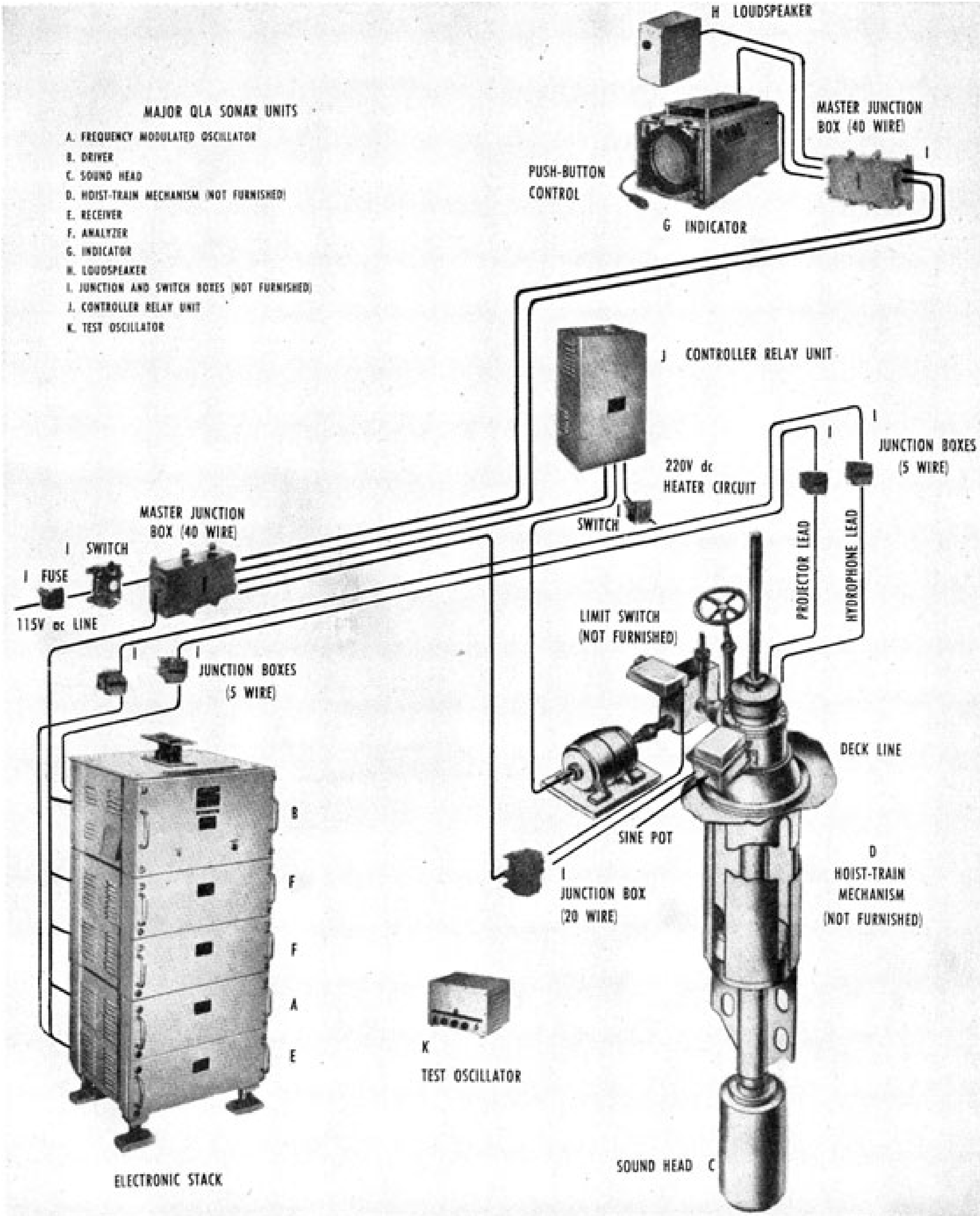


Figure 14-5. -Major units of the QLA f-m sonar.

Range Measurement

During the time required for sound of a particular frequency to reach the target and to return as an echo, the frequency of the projected sound decreases. The longer the travel time, the greater is the decrease and the greater is the difference in frequency between the echo and the sound being radiated as the echo arrives. Figure 5-20 shows that the frequency difference, *f*, between echo and signal is proportional to travel time. It is evident, therefore, that the difference in frequency between a returning echo and the signal being transmitted when the echo is received, is proportional to the *range* of the reflecting object.

The QLA sonar receiver mixes the echo with the signal that is being transmitted and produces a beat frequency equal to the difference in frequency between the echo and the transmitted signal. This frequency difference is presented to the operator both audibly and visually. The audible indication is a musical tone of constant pitch in the loudspeaker; and the visual indication is a spot of light on the cathode-ray indicator.

The difference frequency can be any frequency between 0 and 10 2/3 kc. However, only frequencies between 500 and 2,000 cycles per second are analyzed and used to indicate range. This band was chosen for technical reasons, including ear sensitivity and filter-design considerations.

The analyzer resolves the frequencies between 500 and 2,000 cycles per second by use of 20 bandpass filters, 20 detectors, and an electronic switch. The receiver output is applied to all 20 filters (figure 14-6). The signal at the output of each filter depends on the frequency of the signal and, hence, on the range to the target. The filter outputs are rectified and applied to the intensity amplifier of the indicator. The electronic switch applies the output from filter 1 through filter 20 in sequence as brightening voltage to the oscilloscope. During this time the spot on the oscilloscope is moved radially outward from the center of the screen. Thus, for each of the 20 filters there is a corresponding radius on the screen. For example, a signal with a frequency of 500 cycles per second brightens the trace at a point 3/4 of an inch from the center; a signal with a frequency of 2,000 cycles per second brightens the trace at a point 3 inches from the center.

Range Scale

The range scale is selected by changing the rate at which the f-m oscillator sweeps in frequency. The rate at which the oscillator sweeps determines the frequency difference corresponding to a given range. The greater the rate of change of frequency, the greater is the difference (number of cycles per second) representing a given range.

The oscillator can be swept at five rates. Thus, the operator can select one of five range scales. The rates of sweep are such that range scales of 300 feet, 300 yards, 600 yards, 1,200 yards, and 3,000 yards are available. The periods of sweep corresponding to these ranges are 0.67 second, 2.0 seconds, 4.00 seconds, 8.00 seconds, and 20 seconds, respectively.

Bearing of the Indicator Sweep

The bearing of the hydrophone determines the angular displacement of the sweep on the indicator. As shown in figure 14-6, the hydrophone training mechanism operates through a sine potentiometer and a sweep generator and orients the trace so that the trace is at the same angle (with respect to the vertical) that the hydrophone axis is with respect to the heading of the ship. The indication of any echo on the screen therefore appears in a position corresponding to its relative bearing

Maximum Scanning Rate

The maximum angular rate of speed at which the QLA sonar can scan depends on the maximum range for which the equipment is being operated. The projector transmits sound into the water over an arc of 80°-that is, 40° on each side of the hydrophone. Thus, a particular target is in the field of the projector for about 40° of its rotation before it is received by the hydrophone. The sound head must rotate less than 40° in the' time required for the sound to travel to the maximum range and back.

The maximum useful speed of rotation of the sound head (in revolutions per minute) is approximately 5,000 divided by the range in yards. On short ranges the speed is limited by the characteristics of the electrical system to about 10 revolutions per minute. In a particular installation the choice of speeds is dictated by the service intended. The speed of rotation at long ranges can

266

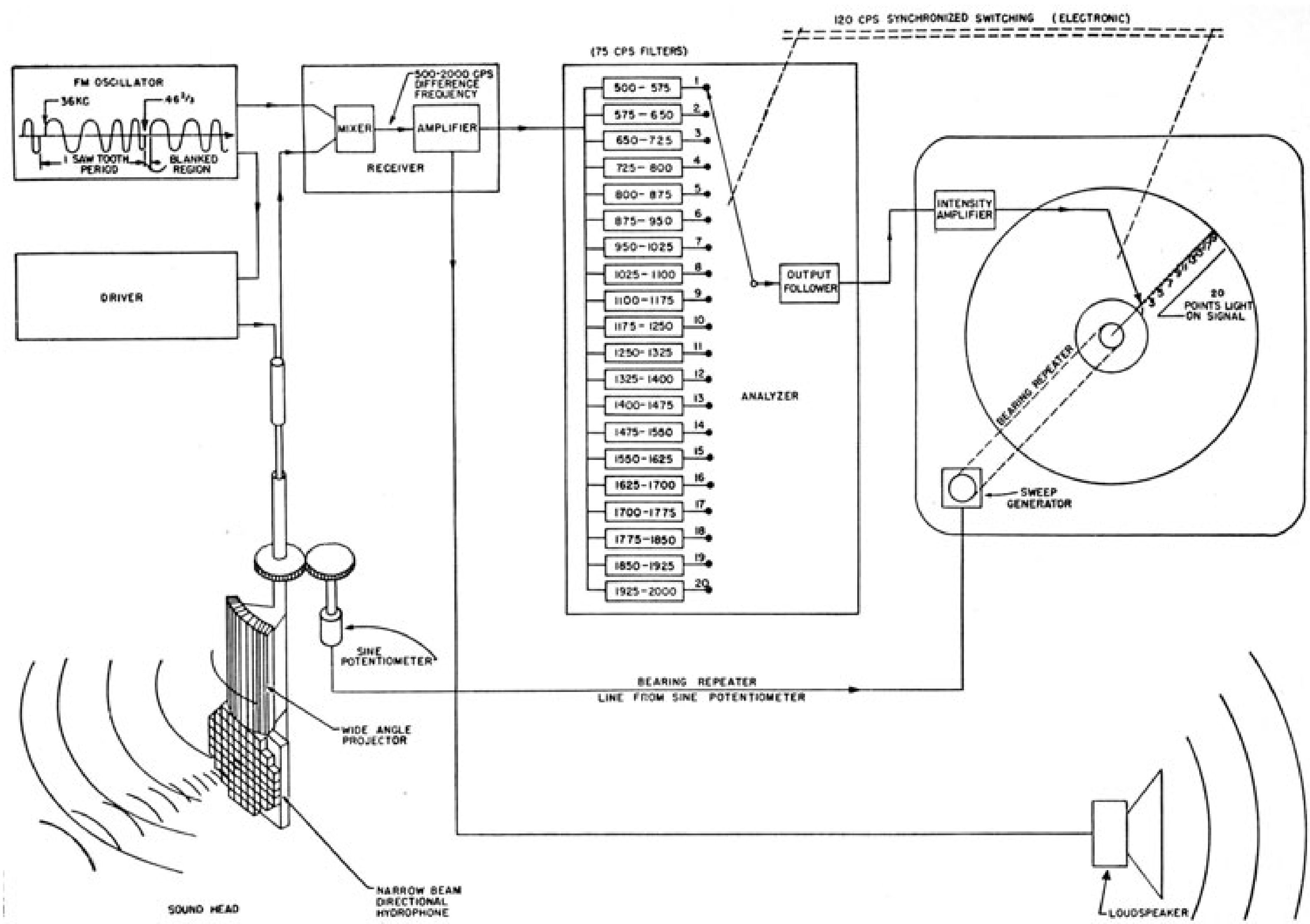


Figure 14-6. -Functional block diagram of the QLA f-m sonar.

267

be increased if the search angle is limited to about 50°; the whole area can then be covered by allowing this search angle to progress from one sweep to the next.

DOPPLER EFFECT

The QLA sonar uses the frequency of an echo in determining range. Therefore, any doppler causes an error in the measurement of range. The error is 75 cycles per second (or one channel on the screen of the cathode-ray tube) for each 2 ½ knots of relative range rate. The error causes the measured range to be too long when closing, and too short when opening, the target. In ranging on moving vessels, the larger part of the echo comes from the wake. The doppler effect in this case is due largely to own ship's doppler.

268

CHAPTER 15

ECHO-SOUNDING EQUIPMENT

The Echo-Sounding Problem

GENERAL

Determining the depth of water to ensure safe navigation is an old problem. The Bible mentions sounding in Chapter 27 of the Acts of the Apostles. The passage concerns the shipwreck of St. Paul, as follows:

"After midnight the shipmen deemed that they drew near to some country. And sounded, and found twenty fathoms; and when they had gone a little further, they sounded again, and found it fifteen fathoms. Then fearing lest they should have fallen upon rocks, they cast four anchors out of the stern, and wished for the day."

It is interesting that the word "sound," meaning to take depth measurement, derives from a different root than the word "sound," meaning a stimulus to hearing, and originally the two meanings of the word had nothing in common. Now, however, the use of echoes for "sounding" has established a new relation between the old meanings of the word.

Until recently the lead line was the primary means of sounding. Today, the lead line is used chiefly to obtain samples of the ocean bottom, because modern methods of sounding with echoes are more efficient. The latest engine-driven lead-sounding machine, employed in some geodetic survey operations, uses a 36-pound lead and is capable of taking a sounding of 300 fathoms in 3 minutes.

The present method of echo sounding is automatic

DEPTH-FINDER REQUIREMENTS

In general, the depth finder should be capable of the automatic recording of the depth in the shortest possible time interval. It must be independent of the ship's speed and must be effective in any depth of water. Depth finders in present use satisfy these requirements. Some of the smaller types do not have a recording unit, but they are useful for small vessels.

The first sonic depth finders used audible sound, which has several disadvantages. Ship noise, which is a maximum in the audible range, causes interference. Because the energy from the transducer cannot be concentrated into a beam, a great part of the energy is wasted. In the sonic range neither piezoelectric nor magnetostriction transducers can be used effectively, but there are no substitutes for them. Ultrasonic sound overcomes all these disadvantages.

Modern depth finders have the following five units in common (figure 15-1):

1. A transducer for the reciprocal conversion of acoustic and electromagnetic energy.

2. An electronic or electric transmitter-rectifier for driving the transducer.

3. An electronic receiver-amplifier for amplifying the weak echo energy picked up by the transducer.

4. An indicator for accurate and continuous

and rapid. A sound is transmitted vertically downward, and the time that is necessary for the sound to travel to the bottom and return to the surface is recorded. Echo sounding is similar to the echo ranging described in previous chapters.

239276°-53-18

indication of depth.

5. A recorder for making a permanent record of water depths over which the ship is passing.

DEPTH-FINDER COMPONENTS

Transducers used in naval sounding equipment are of three types-two of them are magnetostrictive, and the other is a crystal type.

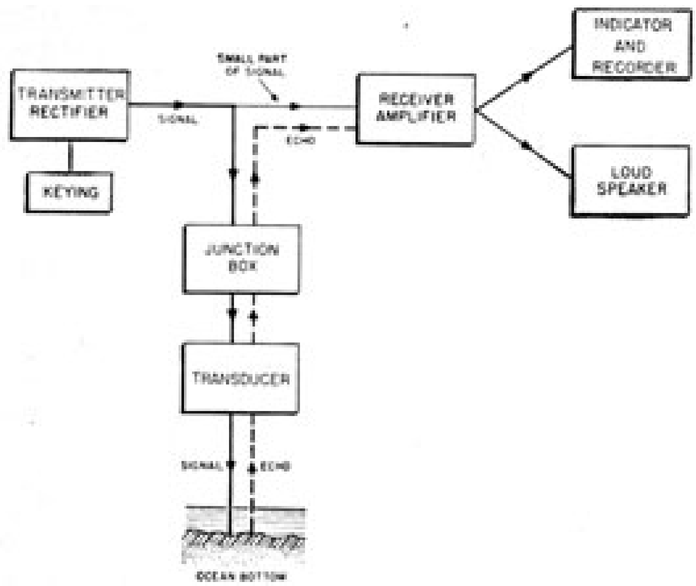


Figure 15-1. -Functional block diagram of a typical echo-sounding system.

The first type, used in large equipment, is similar to that used in echo-ranging equipment (figure 15-2). It consists of a steel plate with nickel tubes mounted on it.

The second type is constructed of nickel laminations. The unit looks somewhat like a transformer (figure 15-3). A current passing through the coil of the winding sets the core into vibration.

The crystal transducers are constructed similarly to echo-ranging crystal transducers except that they are permanently positioned downward.

winding of the transducer, and the output is a damped sine wave.

Indicators are usually the source of the timing impulse to the transmitter. The type of indicators most commonly used consists of a circular rotating disk over a flashing neon light. The type used by the Radio Corporation of America has a hole in a steel tape, which continuously runs on pulleys across the face of the indicator. A long neon light lies behind the tape. The transmitter is keyed as the hole starts over the end of the tube, and the echo returning from the bottom causes the tube to flash. During the elapsed time the hole in the tape has traveled a distance proportional to the depth marked on a scale beside the tape. The tape contains several holes along its length, but only

Transmitters are of two types. The transmitters for the magnetostriction and crystal transducers are in many respects like those used in echo-ranging equipment. They consist of an oscillator and a power amplifier. Transmitters used with laminated transducers are of the impulse type. A capacitor is discharged through the

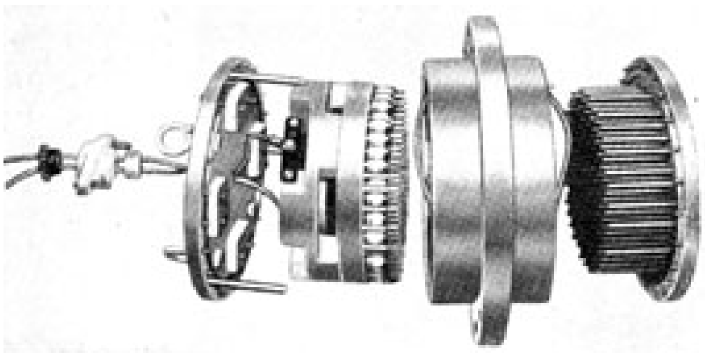


Figure 15-2. -Exploded view of the NMC-2 transducer.

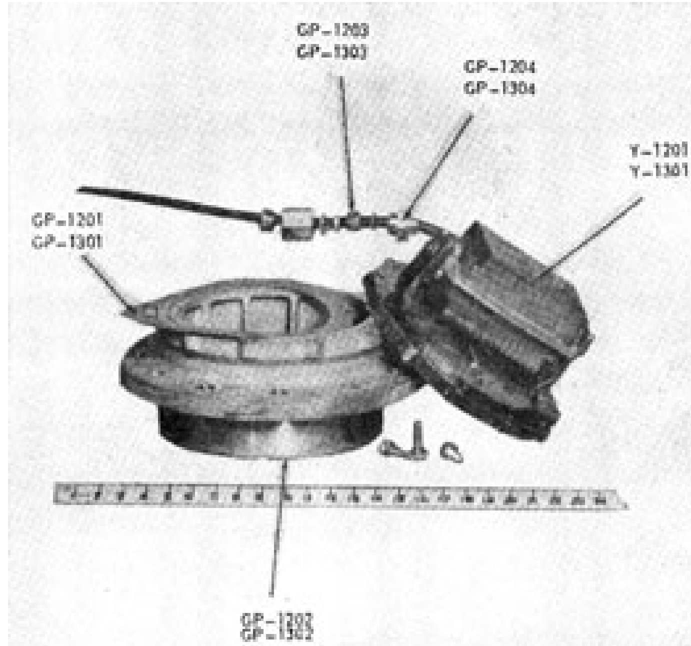


Figure 15-3. -Magnetostriction laminated transmitting projector of the NJ-7 equipment.

one hole appears over the neon lamp at any instant. The tape indicator is usually not so satisfactory as the rotating type. On a small type of depth finder built by Bludworth Marine the depth is indicated on a meter. The most modern type used by Edo Corporation makes use of a circular sweep on a CRT. Depth is indicated by deflections of the sweep.

Depth recorders are all essentially of the same type. They record on a time-depth chart the depth as the time varies. The chart paper is generally of the conducting type with a thin wax insulating coating that is punctured by an electric spark caused by the echo.

NMC-2 Sonar Sounding Equipment

GENERAL

The model NMC-2 sonar sounding equipment is typical of that used on large vessels. It provides a powerful oscillator that allows echoes to be received from depths greater than 2,000 fathoms. A pictorial view of the equipment is shown in figure 15-4. The NMC-2 is similar to the early type of echo-ranging equipment.

The NMC-2 equipment measures ocean depths in fathoms by projecting a signal vertically and measuring the elapsed time before the return of the echo from the ocean bottom. The interval between the emission of the signal and the return of the echo is timed (1) by rotating a disk at a known constant speed and (2) by noting the angular rotation of the disk during the interval by reference to a scale graduated to read directly in fathoms. In depths of less than 2,000 fathoms the operation of the apparatus is automatic, and soundings are obtained with a minimum of attention or adjustment of controls by the operator. A semiautomatic method supplements the automatic method in depths or more than 2,000 fathoms and extends the range beyond that obtainable by the automatic method. The semiautomatic method consists of listening for an audible echo signal in the speaker and noting the white-light position on the proper scale. The NMC-2 is equipped with a recorder that automatically records ocean depths down to 2,000 fathoms.

TRANSDUCER

The NMC-2 equipment uses a magnetostriction transducer of the type shown in figure 15-2. It is mounted near the keel of the ship with the diaphragm horizontal and in contact with the sea water. The normal frequency is 18 kc.

TRANSMITTER AND RECTIFIER

This transmitter is similar to magnetostriction transmitters described in chapter 8. A simplified diagram of the rectifier is shown in figure 15-5. Plate potentials for the transmitter tubes are supplied by the duplex bridge rectifier in figure 15-5. This procedure allows virtually the full output voltage of T406 to be made available to the amplifier. The bridge circuit requires three separate and well-insulated filament windings since they are connected to opposite ends of the load circuit and have the full-load voltage between them.

The plate supply to the oscillators utilizes V407 and V408 with transformer T406 as a fullwave rectifier, and delivers approximately one-half of the total transformer voltage as d-c output.

INDICATOR-RECORDER-AMPLIFIER UNIT

The indicator-recorder-amplifier unit consists of a depth indicator-recorder and a receiver-amplifier with its power supply mounted in the same housing.

Indicator

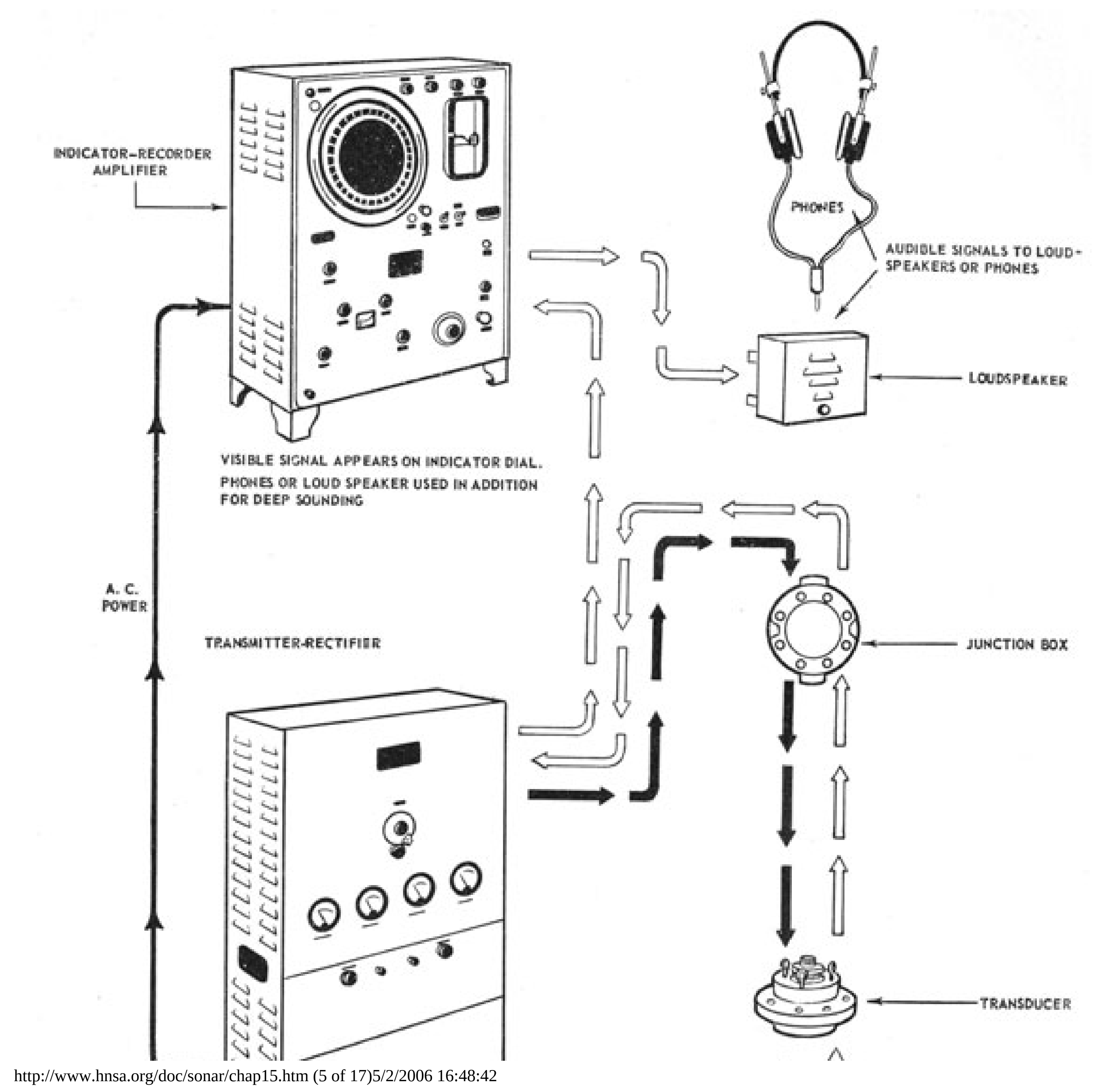
The indicator portion is the principal control unit mounted on the bridge. An *equipment start-stop* button allows the equipment to be turned on from the bridge.

The timing disk rotates behind the two scales, and a slot in this disk is exposed in the space between the *shoal scale* and the *deep scale*. When the visual or automatic method of sounding is used, a neon lamp flashes behind the slot the instant the echo is received. The position at which the light flashes indicates the depth in fathoms. When the audible or semiautomatic method is used, an incandescent lamp shines continuously behind the slot and the operator listens for the echo. The position of the light at the instant the echo is heard gives the depth in fathoms. The position of the *visual-audible* switch indicates which method is in use, whereas the *shoal-*

The transmitter and rectifier unit contains an electron-tube transmitter for generating the alternating voltage to be applied to the transmitting projector, two rectifiers for the high-voltage supply, and two starting relays and switches. A tuning control allows the transmitter frequency to be varied over a range of from 17 to 19 kc.

deep switch selects the scale readings.

In the *400-* or *2,000-fathom* position of the *signal interval* switch an echo indication is obtained at each revolution of the disk, whereas the *800-* or *4,000-fathom* position doubles the depth



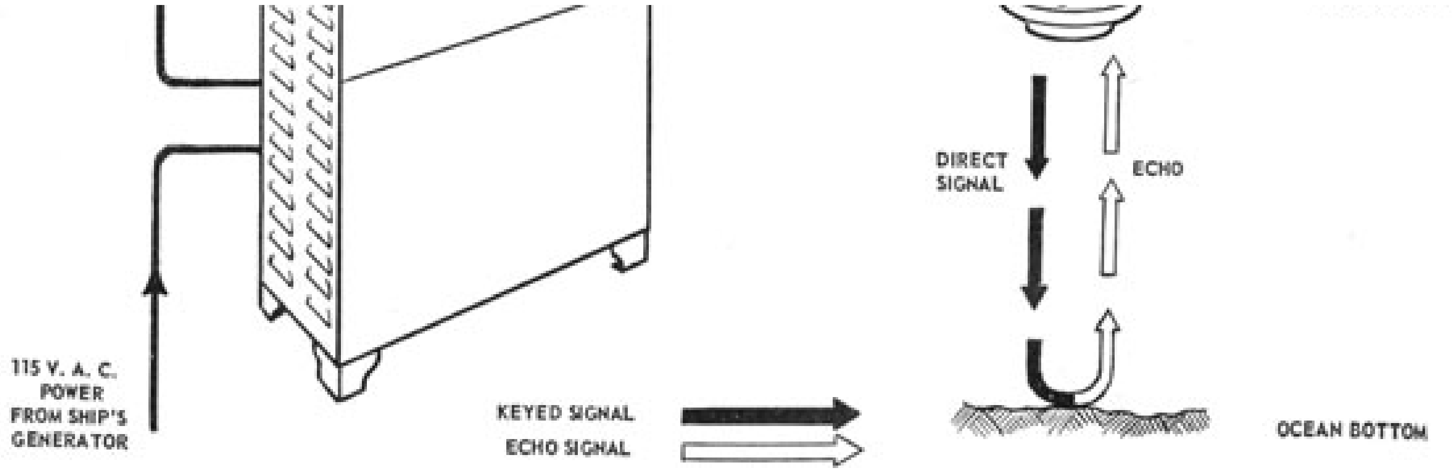


Figure 15-4. -Pictorial diagram of the NMC-2 equipment.

272

---

range obtainable on either scale. The middle position cuts out the automatic keying and permits the use of a manual *test key* in the transmitter. The speaker may be used during visual operation by setting the *visual-audible* switch in the middle position. The middle position also is used for recorder operation.

stylus makes a horizontal mark along the bottom of the chart. This mark shows that the *deep scale* is in use and that the indicated depth must be multiplied by five. Cut-out contacts ground the styluses when they are not over the chart paper. The *chart warning* lamp indicates the need for replacing the chart roll.

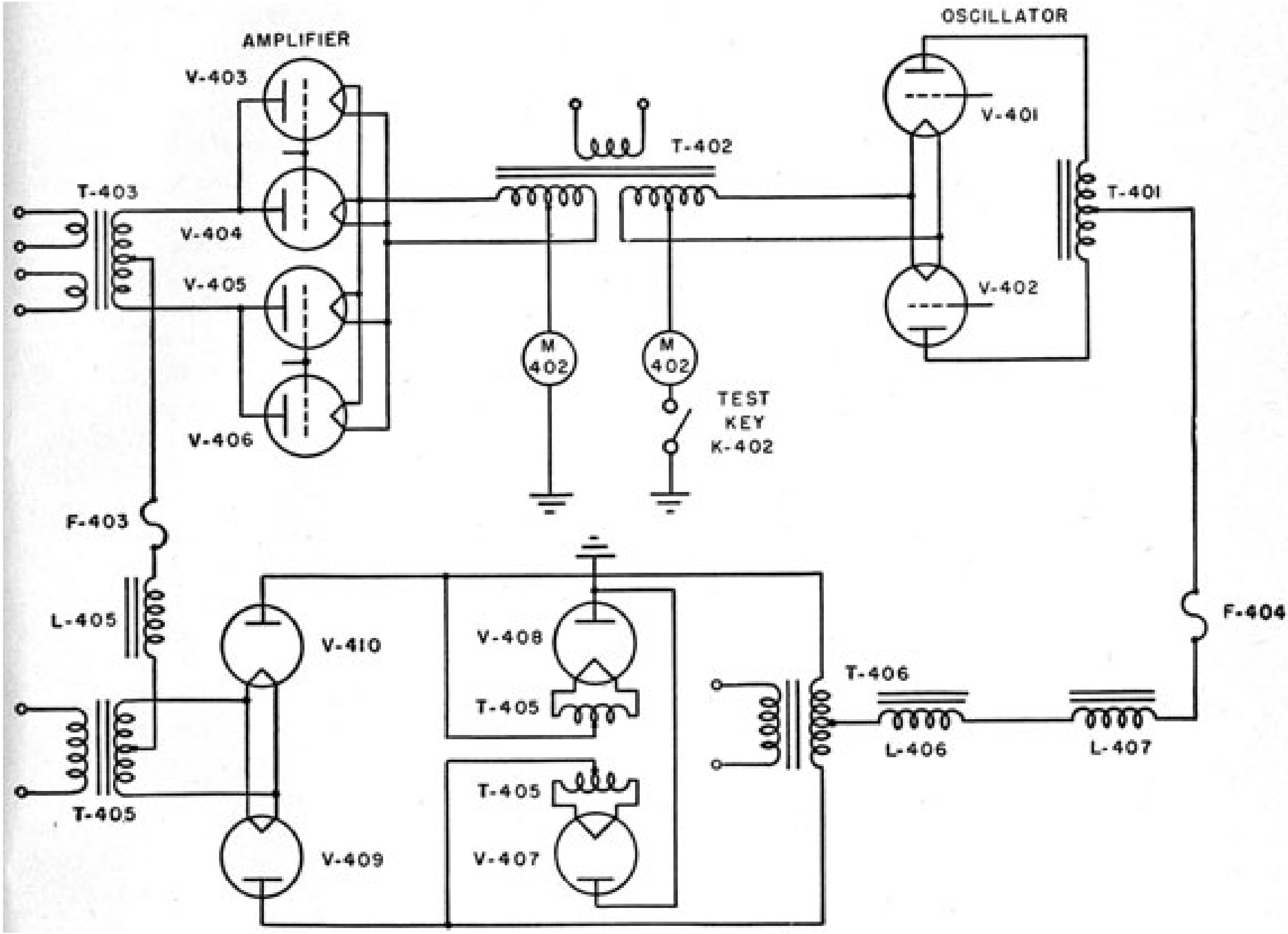


Figure 15-5. -Simplified schematic diagram of the NMC-2 rectifier power supply.

Recorder

The recorder chart is of the dry type, impregnated with conductive material and coated with lead thiosulfate, which turns black upon the passage of an electric current. The stylus produces a mark at the zero line of the chart the instant of signal emission, and the returning echo causes the stylus to produce a second mark at a position corresponding to the depth of water. When the *shoal-deep* switch is in the *deep* position, a fixed

Receiver-Amplifier

The receiver-amplifier unit amplifies the echo to a voltage that is sufficient to operate the red light, speaker, and recorder. It is mounted in the indicator unit and may be adjusted from the indicator panel. Controls include the *tuning* control, which permits the receiver to be adjusted over its calibrated range of from 17 to 19 kc. In addition, there are *volume*, *bias*, *beat note*, and *sensitivity* controls, and a *phones* jack for headset reception.

Rectifier Power Unit

The power supply consists of a vacuum-tube rectifier with a filter circuit to supply the required d-c voltages and a-c filament for the receiver.

TRANSDUCER JUNCTION BOX

The leads from the transmitter to the transducer pass through a junction box. This junction box contains a capacity network for power-factor correction.

NJ-9 Sonar Sounding Equipment

GENERAL

The NJ-9 is an echo-sounding system for use on small craft. This equipment is more compact and is lighter than the NMC-2 equipment, which was designed primarily for ships larger than those of the PC type. All echo-sounding equipment has a transmitter, a transducer, a receiver, and an indicator. The NJ-9 includes (1) a visual indicator of the same type used by the NMC-2 and (2) a recorder for giving a permanent record. This recorder is identical to that used in the NMC-2. The primary difference between the two equipments is in the transducer. The echo-sounding transducer used in the NMC-2 equipment shown in figure 15-2 uses nickel tubes as the magnetostrictive element. The NJ-9 uses the laminated type of magnetostriction transmitting projector (figure 15-3), which is shock-excited by the transmitter and which oscillates at its own natural resonant frequency, thus giving a damped sine-wave output. This type of projector requires a special transmitter to produce a very high current pulse through its winding. The shape of the pulse is not critical and is obtained by discharging a capacitor through the transmitting projector at regular intervals determined by the keying interval.

Figure 15-6 shows the complete schematic diagram of the NJ-9 equipment. The illustration shows that the NJ-9 uses a separate projector and hydrophone.

There are three units and a motor-generator set in

The transmitter functions through the charging and discharging of capacitors. The discharge from one of these capacitors produces ionization within the discharge tube, which then becomes conductive and discharges the power capacitor through the transducer.

When power transformer T401 (figure 15-6) is energized with 115 volts of alternating current the rectifier circuit closes after a 30-second delay introduced by relay K401. This delay permits the cathode of rectifier V402 to come up to operating temperature. The a-c power is converted into high-voltage direct current, which immediately charges capacitor C403. The power-output capacitor, C404, also is charged by this high voltage through resistor R402, which introduces a time constant of 0.02 second. Simultaneously, capacitor C401 is charged through resistor R401 and is discharged through the primary of transformer T402 when the keying contacts of the indicator close. The discharge of current through the primary of transformer T402 is stepped up on the secondary of the transformer to a high voltage, which causes the argon discharge tube, V403, to ionize.

When ionization takes place, the gas in tube V403 offers a low-resistance path for discharging capacitor C404. Capacitor C404 is connected in series with the transmitting projector, and its discharge current must pass through the winding on this projector. Because the resistance of the total discharge path is small, a heavy current flows. This current causes the tube to flash a brilliant bluish-white light.

the NJ-9 system. Because the indicator-recorder used in the NJ-9 system is the same as that used in the NMC-2, it is not discussed further at this time. The other two units are the transmitter-rectifier and the receiver-amplifier.

TRANSMITTER-RECTIFIER UNIT

The transmitter-rectifier unit may be divided into three major electric circuits-(1) rectifier, (2) keying, and (3) discharge.

Immediately after discharging, capacitor C404 acts as a short circuit on the discharge tube, which deionizes because there is insufficient potential across the tube to maintain ionization. Capacitor C404 then is charged from the high-voltage supply and is discharged when the indicator-keying contacts close. This cycle of charging and discharging capacitor C404 is repeated approximately 120

FOLDOUT - Figure 15-6. -Schematic wiring diagram of the NJ-9 sounding equipment.

---

times per minute when the fathoms-keying contacts are operating and 720 times per minute when the feet-keying contacts are operating.

RECEIVER-AMPLIFIER

The receiver-amplifier consists of (1) two stages of amplification (V101 and V102) tuned to the frequency of the projector, (2) a detector (V103), and (3) an output stage (V104). A milliammeter, M101, at the front of the unit indicates the anode current of the output tube and thus provides a check on the operation of the unit. A bias adjustment, R109, operated by means of a screw driver, controls the detector bias and thus the minimum signal level. A power supply, including rectifier tube V105 and its filter system, provides d-c voltages for the anodes and a low a-c voltage for the tube heaters.

PROJECTOR AND HYDROPHONE

The projector and hydrophone are of the magnetostriction type, and each of them consists of an assembly of nickel laminations with an electric winding. The transmitting projector has a low impedance with only a few turns in the winding, whereas the receiving hydrophone is of much higher impedance and has a correspondingly greater number of turns. The frequency of the projector is controlled by its natural period, which is approximately 21.6 kc.

NK-7 Sonar Sounding Equipment

GENERAL

The NK-7 sonar sounding equipment is portable and is designed to operate from a 6-volt battery in boats in which no permanent echo-sounding installation can be made and where the depth to be recorded does not exceed 200 fathoms. The general arrangement of this equipment is shown in figure 15-7.

The NK-7 system is essentially the same as the NJ-9 system, but the NK-7 transmitter power output is not so great. The NK-7 has no visual depth indicator, only a recorder. The projector and hydrophone are of the same type as those used in the NJ-9 equipment. A schematic diagram of the complete NK-7 unit is shown in figure 15-8.

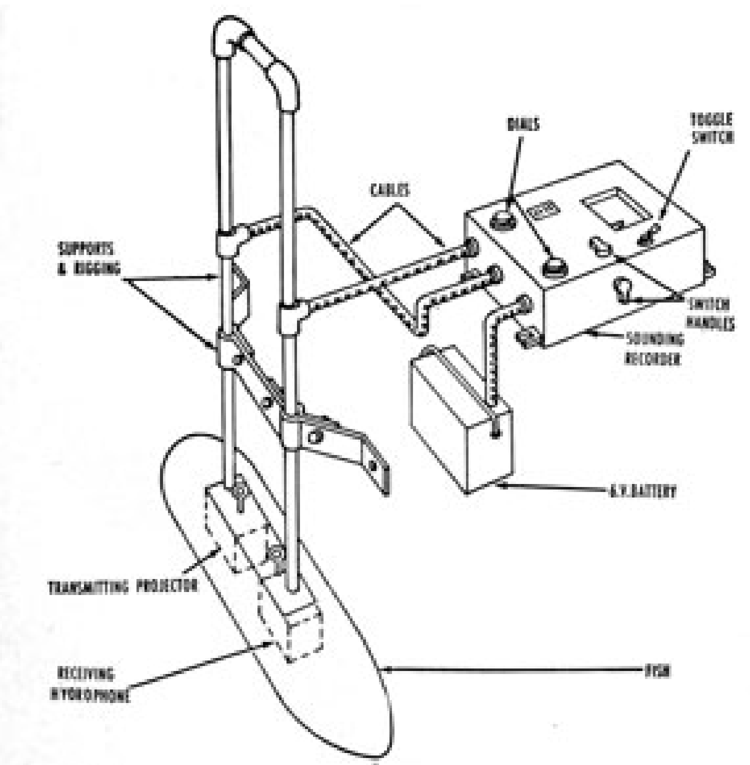


Figure 15-7. -Pictorial diagram of the NK-7 sounding equipment.

RECORDER

S101 is the main power switch. Because of the mechanical arrangement of the main switch, successive turnings alternately close and open the circuit. This switch also reverses the polarity for motor B101 but retains a fixed polarity for the chart lights, vibrapacks, and tube filaments. A specially designed cam attached to the chart feed roll operates switch S105, which keeps reversing the polarity of the input leads to the motor. This action prevents deterioration of the governor contacts caused by electrodeposition of metal from one contact to another. As a result the contacts wear evenly.

This polarity-reversing operation does not affect, the direction of stylus rotation, because the field and armature currents of B101 are both reversed simultaneously. The speed of motor B101 is closely controlled by the centrifugal motor governor. Contacts B101A of this governor are closed normally when the motor is stopped or is running slowly. As the motor speed increases beyond the correct value determined by the adjustment of the speed control, the contacts open because of centrifugal force. This action places resistor R101 in series with both the field and the armature of

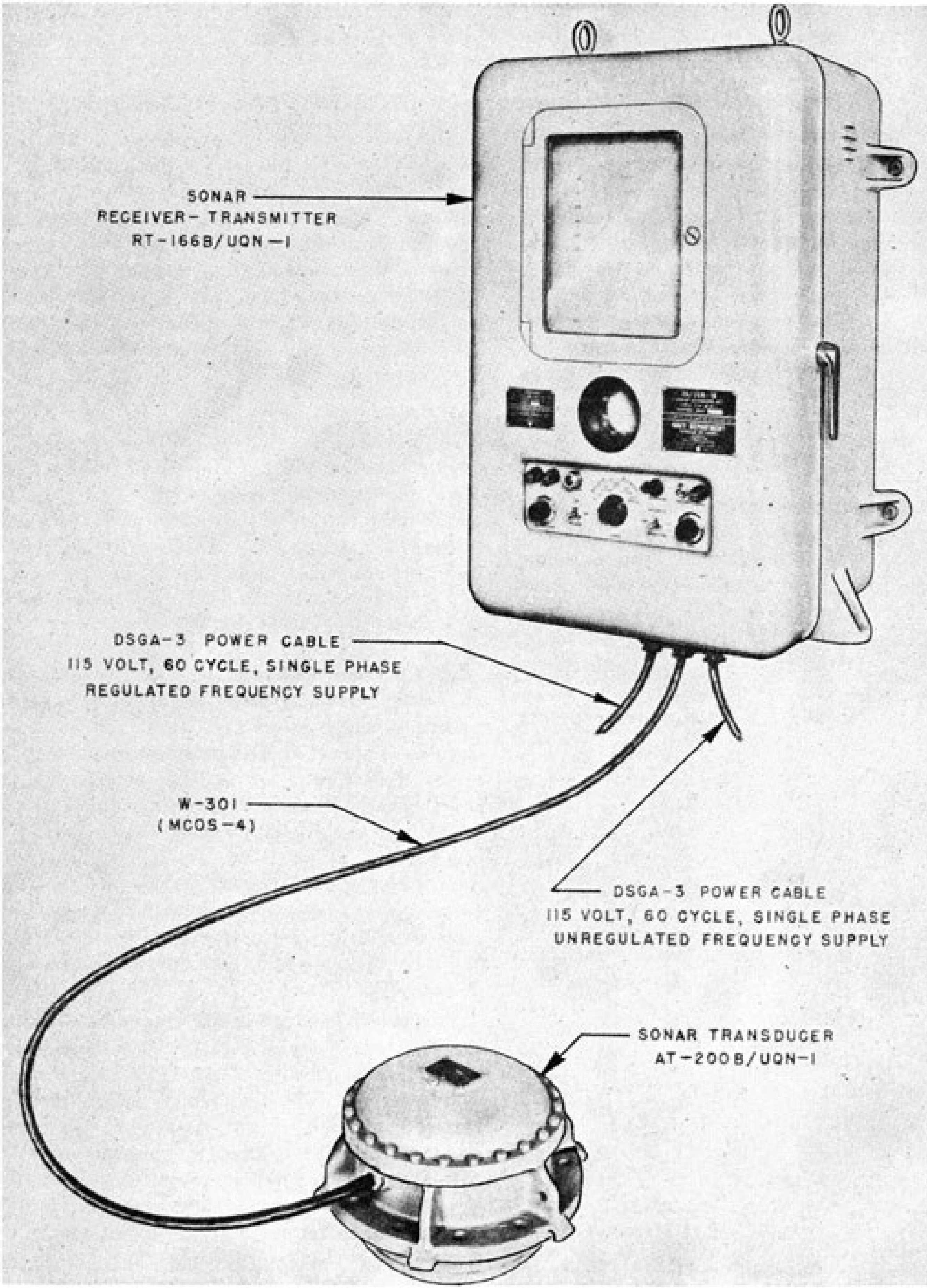


Figure 15-9. -Units of AN/UQN-1B sonar sounding set.

FOLDOUT - Figure 15-8. -Schematic wiring diagram of the NK-7 portable depth recorder.

the motor, thus limiting the flow of current and slowing down the motor.

As the motor continues to slow down, the contacts close; resistor R101 becomes shunted; current increases through the armature and field; and the motor speeds up. Thus, by action of these contacts, the motor attains the correct speed of 4,026 revolutions per minute. The *speed-control* knob permits adjustment of the motor speed by changing the spacing and tension of the governor contact mechanism. Capacitor C103 limits sparking of the governor contacts, which prevents interference with radio reception. Switch S106 is connected mechanically to the *feet-fathoms* lever. This switch is opened only when the lever is in the middle position. This operation cuts down the current to the motor by shunting the supply through resistor R102, and the motor slows down. This slowing action allows easy meshing of the gears when shifting from feet to fathoms.

When closed, the *direct signal* switch, S102, permits direct coupling between the transmitting element and the amplifier through capacitor C101. This coupling allows the emitted signal to be recorded immediately on the chart. This recorded transmitter signal is the zero mark. Motor B102 operates the blower, which provides forced ventilation for the recorder case.

the chart and ground and thus produces the depth record. Resistor R104 limits the current through the stylus needle.

VIBRAPACKS

Power from the 6-volt storage battery is applied to vibrapacks D101 and D102, which in turn provide approximately 300 volts d-c. The output of the vibrapacks supply both the screen and plate circuits for the amplifier tubes. As mentioned previously, D101 also supplies electric power for energizing the magnetostrictive transmitting element.

These power units are complete assemblies. In vibrapack D102 a rapidly vibrating reed interrupts the 6-volt circuit through the primary of power transformer D102H and produces a pulsating direct current. A high-voltage alternating current is thus produced in the secondary. This current is rectified by a mechanical synchronous rectifier arrangement in the vibrator. Choke D102A and capacitor D102D provide a filter for the high voltage inside the vibrapack. Choke D102B and capacitor D102C prevent the ripple voltage from feeding back to the 6-volt line, thus preventing interference.

The operation of D101 is the same as that of D102. The output from both vibrapacks is filtered still further by capacitors C137 and C138.

Transmission of Signal

The electric power for energizing the magnetostriction transmitter is obtained from the 300-volt d-c output of vibrapack D101, which charges capacitor C104 through resistor R103 while the keying contacts, E103 and E102, are open. R103 is a current-limiting resistor inserted to prevent overloading the vibrapack. At the instant contacts E102 and E103 close, capacitor C104 discharges through the windings of the transmitter causing it to vibrate at its natural frequency of about 21 kc. This emitted signal occurs once each revolution of the stylus arm, at the point of zero marking on the chart.

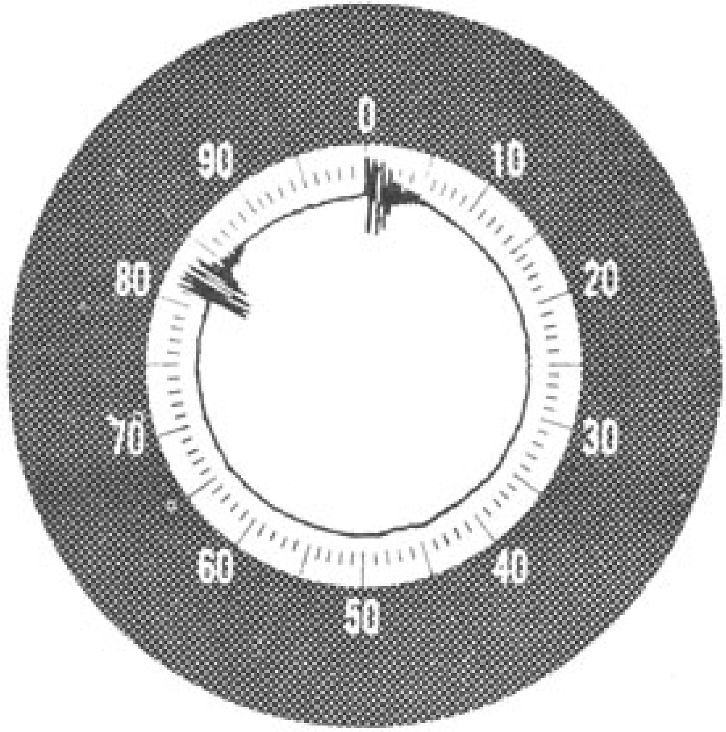


Figure 15-10. -Typical presentation on CRT indicator.

Reception of Signal

In normal sounding, the echo is picked up by the receiver element and, after passing through the amplifier, is led to the stylus assembly in the form of high-frequency alternating current at about 300 volts. This voltage discharges through

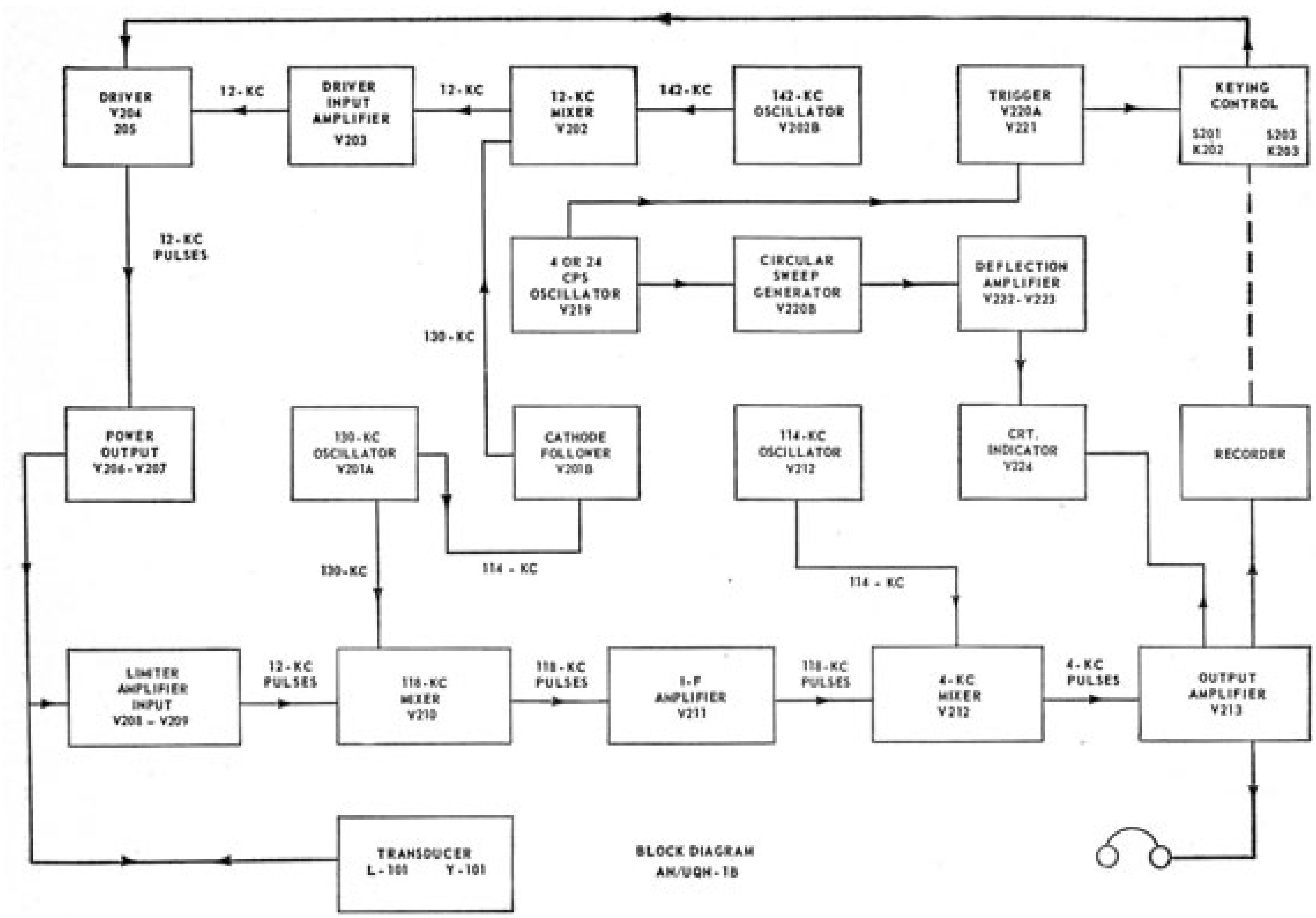


Figure 15-11. -Block diagram of AN/UQN-1B.

The vibrating elements are enclosed in a metal tube having an internal sponge-rubber support, which reduces vibrator noise and insulates the elements from the metal case. Because the metal tube is spun on to a special multiprong plug-in connector, the contacts are not accessible for cleaning or adjustment. If defective, the vibrapack must be replaced with a spare.

AN/UQN-1B Sonar Sounding Set

The AN/UQN-1B is one of the most modern types of sounding equipment being installed on ships today. In contrast with the old types of equipment, the AN/UQN-1B comprises only one small unit with its associated transducer. A photograph of the entire equipment is shown in figure 15-9. In spite of its small size, it gives very accurate readings, at a very wide range of depths, from about 5 feet to 6,000 fathoms.

The equipment is designed for installation on either submarines or surface vessels for the purpose of measuring and either indicating or recording water depths. Three recorder ranges are provided-0 to 600 feet, 0 to 600 fathoms, and 0 to 6,000 fathoms. Two indicator ranges are provided-0 to 100 feet and 0 to 100 fathoms. Means are provided for transmitting a single ping or for automatically keyed operation. The equipment operates by emitting a pulse of ultrasonic energy into the water and measuring the time required for the pulse to travel to the bottom and return.

When recording, a stylus starts across the recorder chart simultaneously with the emission of the pulse. The stylus moves at a constant velocity and marks the paper twice-once at the top of the chart when the pulse is transmitted and again

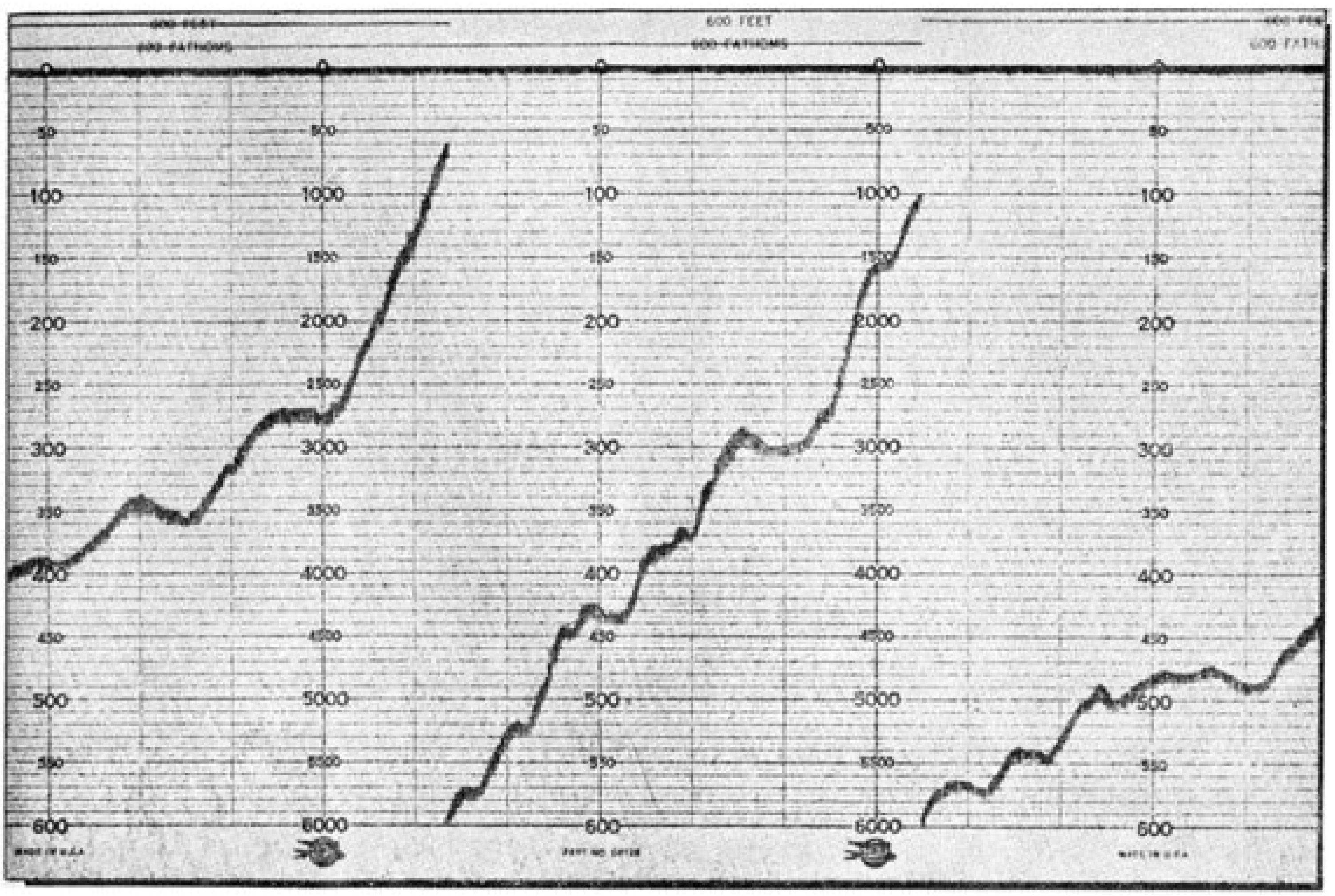


Figure 15-12. -Typical AN/UQN-1B depth recording.

when an echo returns. This procedure provides two points spaced in proportion to the depth of water beneath the transducer. Visual indication is provided by a circular sweep on the face of a cathode-ray tube (figure 15-10). The transmitted pulse and the returning echo radially modulate the trace. An engraved translucent shield in front of the CRT furnishes a scale. The transmitted pulse, which always occurs at zero on the scale, and the echo appear as small radial bars across a luminous circle. The uniform angular velocity of the trace provides the desired time-depth relationship.

The mode of operation is selected by use of the appropriate controls on the front panel of the equipment.

The transducer comprises an array of ammonium dihydrogen phosphate (ADP) crystals in a pressure-tight, flanged housing. It is designed for flush mounting in a standard hull ring of the bottom plating of a surface vessel or outside the pressure hull of a submarine. A tuning inductor is mounted inside the housing. This inductor, with the capacity of the crystals, forms a series-resonant circuit at 12 kc.

The dimensions and arrangement of the crystals and a monel backing plate produce maximum energy transfer at 12 kc.

There are no tuning controls on the equipment because all of the oscillators are crystal-controlled, and all frequencies are fixed. There are three oscillators that provide basic frequencies of 114, 130, and 142 kc. A fourth oscillator provides either a 4- or a 24-cps frequency to supply the circular sweep generator. These oscillators are shown in the block diagram in figure 15-11. From the three basic frequencies, the following resultant frequencies are obtained:

1. 12 kc (142-130 kc) for transmitter operation.
2. 118 kc (130-12 kc) for receiver i-f operation.
3. 4,000 cps (118-114 kc) for chart marking,

CRT modulation, and listening.

The transmitter delivers 800 watts of pulsed 12-kc power through a transmission line to the transducer. The transmitter is a series of transformer-coupled amplifier stages consisting of a single-ended input amplifier (V203), a push-pull driver stage (V204 and V205), and a class B, push-pull, power-output stage (V206 and V207). Transmitter input voltage is constant; keying is accomplished mechanically for recording and with a triggered gas tube for indicating by completing the cathode circuit of the drive tubes, input amplifier, and 130-kc cathode follower.

The receiver takes its input from the 12-kc transmitter pulse, and mixes it with a signal from the 130-kc oscillator, V201A, and the resulting difference signal of 118 kc is used as the i-f frequency. This 118-kc signal is amplified and mixed with a 114-kc signal to produce the 4,000-cps audio frequency.

The recorder is conventional and has three ranges. An actual recording sheet of this equipment is shown in figure 15-12. This equipment will perform well at any depths encountered.

CHAPTER 17

SONAR TRAINING EQUIPMENT

QFA-6 Attack Teacher

DESCRIPTION

The model QFA attack teacher is a training device that provides operational instruction in the use of searchlight types of sonar echo-ranging and listening equipment and in under-way control of the motion of vessels engaged in antisubmarine warfare. The QFA equipment is installed normally on shore stations.

As shown in figure 17-1, the QFA-6 equipment consists of an optical projector, a ship steering stand, a ship sonar console, a sound-range recorder, a submarine steering stand, a submarine sonar stack, a screen, and an attack-aids adapter. The screen corresponds to a miniature ocean, on which are projected miniature images of a ship and a submarine. These images can be maneuvered independently by remote control from the steering stands. The submarine image can be maneuvered also by controls on the projector. The ship image, however, must be maneuvered from the ship steering stand. The ship steering stand and sonar console, which are functional counterparts of real equipment, are placed in another room so that the trainee can train his sonar beam, can hear the sonar echo, and can maneuver his ship without seeing the problem on the screen.

The projector contains projecting systems, sound-effect circuits, and submarine-maneuvering controls. The projecting systems project the ship and submarine images, the sweeping sound beam, and a true-bearing line. The sweeping sound beam is a band of light that moves away from the ship and simulates the active area of a real sonar acoustics of a real situation. These circuits simulate target echo, reverberation echoes, and transmission-signal and water noise. The ship's propeller sounds originate in the submarine sonar stack and steering stand. The submarine-propeller sound originates in the sonar console.

The ship steering stand and sonar console are in a ship-control room out of sight of the projector and are operated by a team of trainees. The team in the ship-control room consists of a conning officer, sonarman, tactical range-recorder operator, and helmsman. The sonar console is a counterpart of a real console and has a BDI display that indicates in accordance with the situation portrayed on the ocean. The attack teacher can be operated by one trainee team in the ship-control room and one man at the optical projector. The man at the optical projector resets the problem, maneuvers the submarine, and keeps the true-bearing line pointed at the ship. The attack teacher also can be operated with the submarine steering stand and submarine sonar stack in a separate room with a submarine trainee team. This team then maneuvers the submarine out of sight of both ocean screen and ship stand. An optical-projector man is always required for operating the true-bearing line.

The attack-aids adapter is a unit that simulates the functions of the dead-reckoning analyzer (DRA). When a dead-reckoning tracer (DRT) and an attack plotter (AP) are used in a simulated CIC the attack-aids adapter provides the signals to the tracer and the plotter. The adapter (1) receives ship's-heading and ship's-speed signals from the steering stands, (2) extracts the east-west (E-W) and north-south (N-S) components of motion, and (3) develops step-motor

transmission. The true-bearing line is a line that originates at the submarine and is manually pointed to the ship at all times by an operator stationed at the projector. The line is needed in the simulation of BDI, RLI, and pattern directivity. The sound-effect circuits simulate the signals for driving the DRT and AP.

The antisubmarine-warfare situation is made as real as possible for the sonar operator and

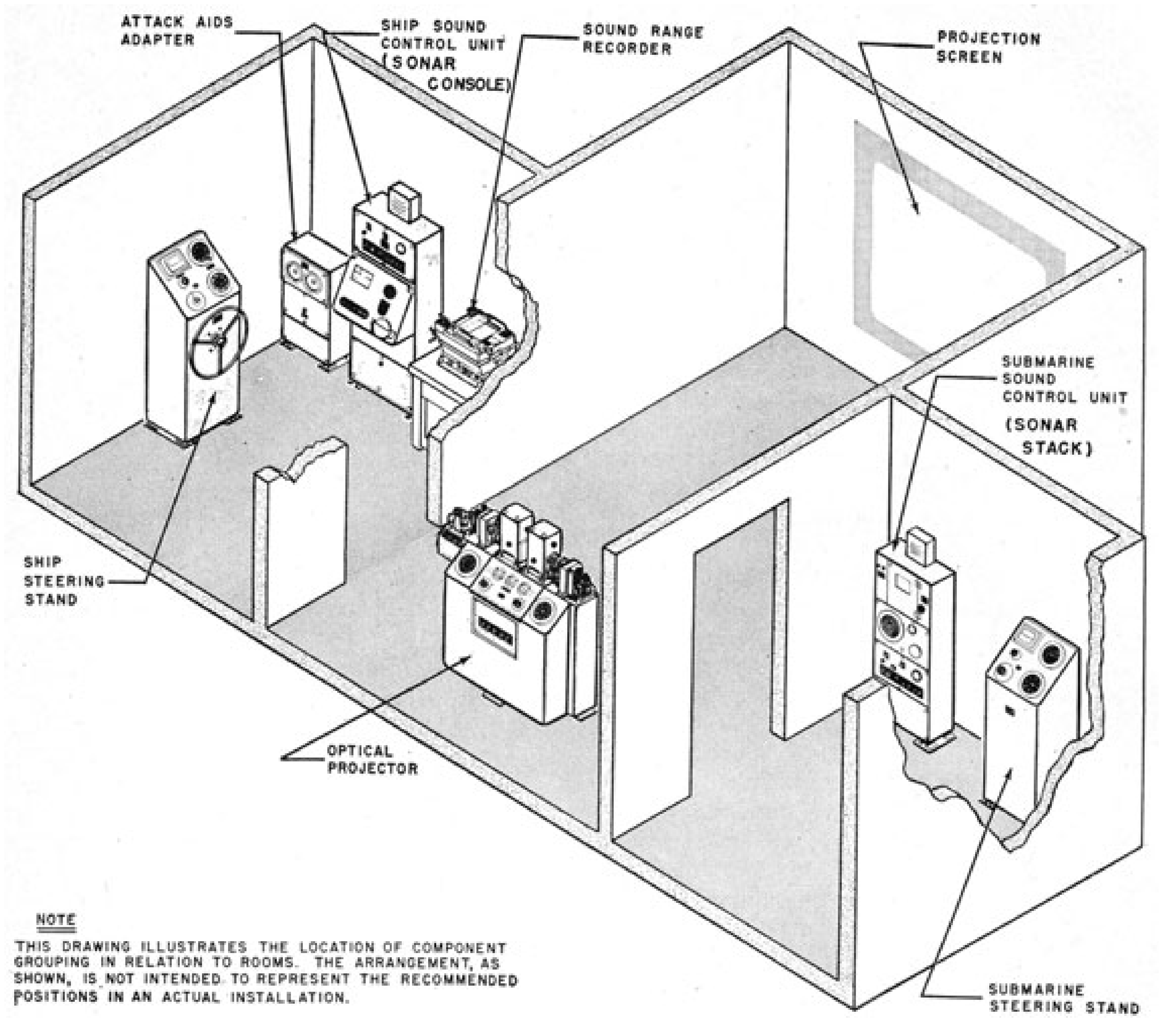


Figure 17-1 -Components of the QFA-6 attack teacher.

prospective conning officer. The images move in accordance with orders from the control equipment. Control-equipment orders are fed manually into the attack teacher by counterparts of real equipment-the sonar console and the ship's helm and engine telegraph. Although the acoustic sounds are simulated by thyratrons, oscillators, interrupted light beams, variacs, watt-hour meter motors, and photoelectric detectors, the sonar console and stack are functionally precise counterparts of sea-going equipment. The attack teacher thus trains officers and operators in the technique of antisubmarine warfare. The operation of the equipment reduces to two fundamental problems in synthesis-(1) control of image motion in accordance with control-equipment orders, and (2) simulation of the acoustics of a real situation.

IMAGES AND THEIR MOTION

The "Ocean"

The "ocean" of the attack teacher is a 50-inch square screen onto which the images representing the submarine and the ship are projected from the optical projector. The screen represents a square section of ocean, the top being north and the right

side being east. The images of the vessels are boat-shaped objects, which rotate as the vessels change heading, thereby indicating to an observer at the projector the direction of vessels' motion. The scale of the ocean varies from 40 to about 170 yards per inch, depending on the projection distance.

An operator stationed behind the screen can plot the positions of the images with grease pencils, thus providing a plot of the courses of the vessels depicted.

Optical System

Two completely independent optical systems produce ship and submarine images. The ship image originates in a projecting system at the right of the projector, the optical axis of the system being parallel to the screen. The light from this projector is deflected vertically by a rotatable first surface mirror, which has its axis of spin perpendicular to the plane of the screen. A second rotatable first surface mirror, arranged to receive the light from the first mirror, has its axis of spin parallel to the plane of the screen and diverts to The submarine image is projected onto the screen in an identical manner. This image originates in the optical system at the left of the projector. Both ship and submarine images can be moved to any point within the confines of the screen. The two images can be differentiated by color and by length, the ship image being green and approximately 120 scale-yards long and the submarine image red and 80 scale-yards long.

The motion of the projecting mirrors is controlled by driving each mirror through gears with a specialized type of induction motor, which is closely related in design and performance to the conventional a-c watt-hour meter motor. Because watt-hour meter motors are used throughout the attack teacher, a working knowledge of the induction watt-hour meter motor is essential for understanding the speed and direction controls of the ship and submarine images.

Watt-Hour Meter Motor

The watt-hour meter motor functions as a split-phase induction motor. It consists of an electromagnet, a rotating element, and associated damping magnets. The electromagnet is composed

the screen the light that is incident upon it. The result of this combination is that (1) rotation of the first mirror produces lateral, or east-west, motion of the image on the screen and (2) rotation of the second mirror produces vertical, or north-south, motion on the screen. Rotation of the mirrors in accordance with the north-south and east-west components of the velocity of the ship results in motion of the ship image on the screen.

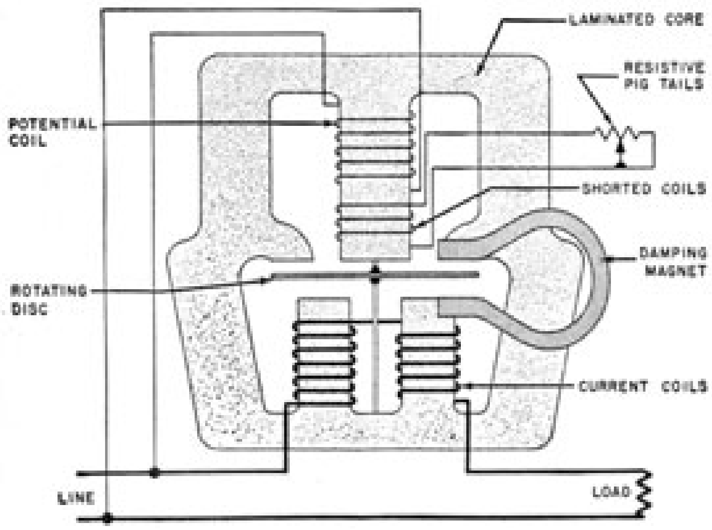


Figure 17-2 -Schematic diagram of a single-phase watt-hour meter motor.

of a potential coil and two current coils. As the names imply, the potential coil is across the line and the current coils are in series with the line (figure 17-2).

The coils are mounted on a common laminated iron core. Their physical relationship is shown in figure 17-2. An aluminum disk is mounted between the potential and current coils on a vertical shaft set in jeweled bearings. This disk is the rotating element.

Because the current coils are in series with the line and carry the load current, they are wound with a few turns of heavy wire. The load current through these coils produces a flux that is proportional to and in phase with the line current.

The potential coil is a high-impedance winding composed of a great many turns of fine wire. The current through this coil is nearly 90° out of phase with the applied potential. However, the currents in the potential and current coils must be in exact quadrature if the speed of the motor is to be proportional to the power factor. To shift the flux of the potential coil so that it is exactly 90° from the flux of the current coils, a small coil, short-circuited through resistance-wire pigtails, is placed in the flux path of the potential coil. The current

---

induced in the shorted coil constitutes a magnetomotive force which combines with that of the potential coil to produce the potential coil flux. By adding the proper amount of resistance to the coil by means of the resistive pigtails, the flux from the potential coil can be made exactly 90° from that of the current coils.

These two quadratured flux components induce eddy currents in the portion of the disk that is in their respective field. The interaction of the eddy currents in the disk and the field across the disk meter motor can be made to rotate in accordance with the east-west component of vessel motion when (1) its current coil is energized with current proportional to vessel speed and (2) its potential coil is energized with a signal proportional to cos θ_2, where θ_2 represents heading away from the east. Because θ_1 and θ_2 are 90° apart, cos θ_2 equals sin θ_1.

The attack teacher uses two watt-hour meter motors,

causes the disk to rotate. The direction of rotation is controlled by the polarity relation of the potential and current coils. Reversing the connections of the potential coil reverses the direction of rotation.

When the motor is operating properly, the torque on the disk is zero for a zero power factor load and greatest for a unity power factor load. For a given set of values of voltage, E, and current, I, the torque is proportional to the load power factor, $\cos\theta$. To calibrate the mechanical output of the motor and to make the motor speed constant for given values of $EI\cos\theta$, damping magnets are mounted so that the disk cuts their magnetic field. The eddy currents thus induced tend to oppose the rotation of the disk. The damping action is proportional to the speed of the disk-it is small when the disk rotates slowly and large when it rotates rapidly. For any given load the driving torque causing the disk to rotate is balanced by the damping action of the drag magnets and the speed is constant. The rotational speed is proportional to $EI\cos\theta$. Because $EI\cos\theta$ is the true average power of the electric circuit, the speed of the disk is a measure of the power being supplied to the circuit.

Mirror-Drive Motors

In the attack teacher one mirror reproduces north-south motion of a vessel and another mirror reproduces east-west motion. Because a watt-hour meter motor moves its rotating disk at a speed proportional to $EI\cos\theta$, one meter motor can be used to extract the north-south component of vessel motion by energizing (1) the current coil of the meter motor with a current proportional to ship speed, and (2) the potential coil with a current proportional to $\cos\theta_1$, where θ_1 represents the heading away from north. Similarly, a second called *coordinate motors*, to reproduce the motion of each vessel. The current coils of the motors are in series and are energized by the same current, which is proportional to ship speed. The potential coil of one motor is energized by a signal proportional to $\cos\theta$; the potential coil of the other motor is energized by a signal proportional to $\sin\theta$, where θ equals the heading of the ship.

Figure 17-3 shows the schematic diagram of the speed and direction controls of the attack teacher. It shows (1) the circuits of the ship controls in the ship steering stand and (2) the circuits of the submarine controls in the optical projector.

N-S and E-W coordinate motors, K1401 and K1402, are single-phase watt-hour meter motors. The current circuits of the E-W and the N-S mirror-drive motors of the ship projection system are in series. The current in these circuits is varied so as to be proportional to the speed of the ship. Thus, the current coils of both coordinate motors receive the same current, which is proportional to ship speed.

The potential circuits of the motors are excited from the secondary of a two-phase phase-shifting transformer, which is positioned as ship's heading. The primary of the transformer is excited from a two-phase generator that is provided with the equipment. The phase-shifting transformer has a rotor similar to that of a two-phase wire wound induction motor rotor and may be rotated to any angular position. As the rotor is shifted the phase angle between secondary and primary voltages is shifted uniformly.

The phase angle of the common current in the current circuit of the coordinate motors is constant with respect to the primary excitation of the phase-shifting transformer. If the phase angle of N-S motor potential with respect to this current is the angle 0, and if the output potential of the phase-shifting transformer is constant, the

FOLDOUT - Figure 17-3 -Schematic diagram of speed and direction controls of the attack teacher.

resultant torque of the N-S element, $L_{N\text{-}S}$, may be expressed as follows:

$L_{N\text{-}S}=k_1 I \cos\theta$.

The potential on the east-west element is advanced 90° electrically in phase, and the torque of this element, $L_{E\text{-}W}$, may be expressed as follows:

$L_{E\text{-}W} = k_1 I \cos(\theta - 90°) = k_1 I \sin\theta$.

Thus, the coordinate motors move at a speed proportional to the E-W and N-S components of the motion of the vessel.

Both mirror motors are equipped with conventional watt-hour-meter motor-damping magnets of such strength that the rotor speed is directly proportional to the torque if the mechanical load on the motor is negligible or compensated for. As previously defined, the current in the motor elements is proportional to the speed of the ship. The equations of torque therefore reduce to the following:

N-S speed= ship speed X $\cos\theta$,
E-W speed= ship speed X $\sin\theta$.

If by calibration, θ is made the true-compass course of the ship, and if ship's heading is maintained thereafter as the angular position of the rotor of the phase-shifting transformer, the mirror speeds are as follows:

The N-S mirror speed is proportional to ship *change* of ship's heading for any given rudder angle must be directly proportional to the speed of the ship. This tactical consideration is injected into the attack teacher by making the *rate of turning* of the rotor of the ship's-heading phase shifter proportional to the current in the mirror-motor circuits.

Other tactical considerations are (1) the acceleration or deceleration delay, which accounts for the time necessary to get a ship to the desired speed, (2) the turning delay, which accounts for the *advance* (the distance traveled before the rudder takes effect) and the *transfer* (the additional distances necessary to enter a constant turning circle), and (3) the loss of speed in a turn. These considerations are injected into the attack teacher by controlling the response of (1) the *rudder motor*-a watt-hour meter motor that drives the rotor of the phase shifter-and (2) the current in the current coils of the coordinate motor. The ship rudder motor, B705, is shown at the left of figure 17-3. The submarine rudder motor, B210, is shown at the lower right of figure 17-3.

The ship steering stand has counterparts of an engine-room telegraph and a speed indicator. As shown in figure 17-3, these units control variacs and watt-hour meter motors, which in turn control the response of the rudder motor and coordinate motors. The rudder-motor positions the rotor of the ship's-heading phase shifter, which in turn determines the position of the image on the screen. The current in the current coils of the coordinate motors determines the speed of image motion.

The engine telegraph operates a ship-speed control through a time-delay circuit, which provides acceleration rates typical of the class of ship depicted. The helmsman's wheel on the steering

speed times the cosine of ship's heading, and the E-W mirror speed is proportional to ship speed times the sine of ship's heading. An identical analysis is applicable to the motion of the submarine.

Tactical Considerations

The control of the motion of the ship and the submarine reduces to control of (1) the angular position of the rotor of the phase-shifting transformers, and (2) the proper variation of the current in the current circuits of the mirror motors. These variables are representative of the direction and speed of the vessel depicted. It is necessary that the tactical characteristics of the vessels represented be as close as possible to the characteristics of real vessels.

One characteristic of a given class of vessels is that the turning circle for any given rudder angle is nearly independent of speed. This characteristic exists because there is little sideway slippage when a ship is in a turn. Therefore, the *rate of* stand actuates another time-delay circuit, which provides rudder delays. This circuit uses the turning-delay variac. The turning-delay variac and rudder variac, which operate the rudder motor, are energized by the engine-telegraph variac so that the turning rate is proportional to ship speed. Therefore, the turning circle of the ship is constant at all speeds below 20 knots-as it should be. Above 20 knots, however, the turning circle increases with speed because the watt-hour meter motors cannot be controlled over so wide a range of speeds.

The loss of speed in a turn is provided for in the rudder-control mechanism by the operation of the speed-decay variac, T705. This variac causes the ship image to slow down when in a turn with 15° or more of rudder angle. The rate of deceleration is determined by the speed-control unit, consisting of a speed-control variac and a speed-control motor. The final value of speed in a turn is determined by an adjustment in the steering stand. A gyrocompass repeater card, which is attached to the end of the rotor of the phase-shifting transformer, indicates the ship's heading.

The controls for the speed and turning of the submarine image are fundamentally identical with those for the ship image. In normal use, the submarine image is controlled from a steering and (2) the sound information as radar range and bearing. This important function should not be neglected by training activities.

SIMULATING THE SONAR SOUND BEAM

In a real situation, the sonar operator aboard ship trains his sonar beam to obtain the range and bearing of the target. Because the images on the attack-teacher screen do not have transducers, it is necessary to simulate the sonar beam and the directional pattern of the beam.

Active Area

In addition to the ship and submarine images and the true bearing line a fourth image on the ocean screen

stand, which is almost identical with that provided for the ship image. As has been mentioned, control of the submarine image is available also at the projector. The fundamental control principles for the submarine image are identical with those for the ship image except that the turning circle of the submarine image is constant at speeds up to 12 ½ knots and increases in proportion to the speed above 12 ½ knots.

The value of the attack teacher for teaching under-way control of vessels is increased by providing for the simulation of the tactical characteristics of many classes of vessels. The variable-speed motors of the rudder- and speed-control circuits can be adjusted over a wide range of speeds. It is possible, therefore, to adjust the equipment to the exact characteristics of any class of vessel by a suitable combination of turning delay, turning rate, acceleration characteristics, and speed loss in a turn. As new surface and underwater vessels are developed and the tactical characteristics are made available, attack teachers must be calibrated accordingly. Therefore, the personnel who maintain attack teachers must understand thoroughly the basic principles and adjustments that determine the tactical characteristics of both the ship and the submarine.

Although the attack teacher is primarily a sonar training equipment, the realism of ship response to the helm and engine-telegraph orders makes the attack teacher useful for under-way ship-handling problems. For instance, it may be used for station-keeping and station-changing problems by using (1) the submarine image as the guide vessel represents the active area of the echo-ranging sound beam. The ship projecting system is equipped with a second optical system, which projects a band of light onto the screen (figure 17-4). The band of light periodically travels away from the ship across the screen at a speed equal to one-half the scale velocity of sound in water. This speed makes the band represent the active area of a real sound beam. The direction of motion away from the ship is controlled by a training mechanism on the sonar console. The zero, or pivoting, point of the beam is coincident with the image of the ship, irrespective of the position of the ship on the screen. The two images optically converge upon each other at the screen and move over the screen together by the rotation of their common mirror system. The traverse of the beam from the zero position is initiated by the keying of the sound equipment. Although all the projected beams originate in d-c excited lamps so that there is no 60-cps modulation of the beams, the light beam that simulates the sound beam is interrupted so that it can be detected electronically.

The submarine projection system has a telescope that is trained on the submarine image through a common mirror system. At the focal point of this telescope is an orifice, which permits only light that is incident on the submarine image to pass through it. An extremely sensitive phototube coupled to a suitable amplifier is placed beyond the orifice of the telescope. A simulated echo signal is produced by the phototube when the sweeping sound beam passes over the submarine. The "echo" amplifier is sensitive only to a-c signals

from the phototube. Therefore, the amplifier is insensitive to the light from the submarine image-which is derived from a d-c source-and responds only to the "a-c" sound beam.

In a real sonar equipment, an ultrasonic wave is transmitted and the resulting ultrasonic echo is heterodyned to an audio frequency. In the attack teacher ultrasonic frequencies are not used. The d-c-excited source that develops the active area of the searching sound beam is interrupted at an audio frequency so that the output of the photo-electric detector (corresponding to the echo signal) is an audio signal (corresponding to the audio output of the sonar receiver).

The sound-beam projection system is equipped with a motor-driven disk with peripheral holes, which interrupt the d-c-excited light at a frequency of 800 cycles per second ± doppler. Therefore, if the sound-beam image crosses the submarine image in its transit across the screen an a-c signal of the frequency of the pulsating light is delivered to the amplifier by the phototube. This amplifier transmits the signal to the sonar console, where it may be both heard over the loudspeaker and seen on the range-recorder trace. The sound beam moves away from the ship at half the scale velocity of sound, and the 800-cps tone is heard the instant the beam reaches the submarine. The elapsed time is the same as if the beam traveled at the scale velocity of sound and were reflected from the submarine back to the ship before being detected. Therefore, the range-indicating or recording equipment indicates the true-scale range. The bearing of the target is determined by the rotatable angular position of the axis of the sound beam. The bearing is indicated by conventional bearing repeaters.

Slant Range

In echo ranging, recorded or indicated range is complicated because the target usually is below the surface of the ocean. Therefore, measured range in a real sonar is the slant range or distance to the target. The range across the surface of the ocean to a point directly above the target is defined as the *horizontal range*. Most of the interpretive devices employed in attack procedures assume that the recorded or indicated range is identical with the horizontal range. With very deep targets, a substantial error is introduced by the discrepancy between slant and horizontal range because the discrepancy increases as the attacking ship moves close to the target. This practical difficulty has led to the inclusion in the attack teacher of a means for producing the slant-range effect. The sound-beam projecting device can be modified to provide simulation of the slant-range effect for any desired target depth. This modification is accomplished by substituting a cam in the projecting mechanism. Five arbitrary target depths are provided-0, 100, 200, 300, and 400 yards. A different cam must be substituted for each depth.

Sound-Beam Training Control

The sound-beam training-control circuit can be operated manually or by either or both of two additional control circuits-(1) maintenance of true bearing (MTB), which maintains the sound beam on a constant true bearing regardless of changes in the vessel's heading, and (2) automatic search, which provides variable search programs without operator

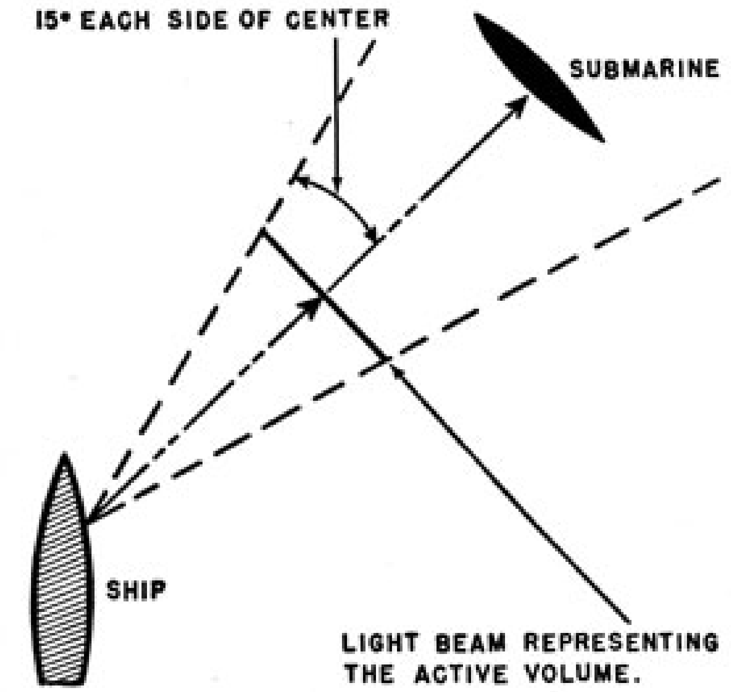


Figure 17-4 -Projected band of light that simulates the active area of the sound beam.

control.

The maintenance-of-true-bearing circuits are the same in principle and in function as those of standard equipment. A switch representing a battle-damage switch provides for relative-bearing training procedures.

The automatic-search provisions in this equipment depart from existing standard equipment in

that the continuous-rotation principle is employed. A mechanism is provided whereby the true bearing of the sound beam is changed continuously at three discretionary speeds. The sonar equipment is keyed at each 5° of train, thus providing a search pattern of 5° regardless of the keying interval. In other words, the keying interval is fixed by the rate of rotation of the sound beam.

The bearing indicator associated with the training-control unit is entirely different from that of standard equipment. The conventional azimuth ring, rotating lubber line, and inner compass card are replaced by (1) an edgewise card indicating true bearing and (2) a small relative-bearing indicator, which may be read only to within 5°.

The QFA-6 uses a beam that starts at the ship. Therefore, only the ship station has simulated sound-ranging equipment. Range information is provided to the submarine station by using the ship returns, as will be explained later.

the true bearing of the ship from the submarine (or of the submarine from the ship) is available for control purposes.

The true-bearing line is maintained in proper position-that is, pointing from the submarine image to the ship image-by the optical-projector operator. The bearing of the true-bearing line determines the position of (1) a vane in front of a photoelectric detector, and (2) a rotor of a phase shifter, called a signal splitter. Whenever the sound beam falls exactly on the true-bearing line, the *signal splitter* sends a no-deviation signal to the BDI. When the sound beam is to the right or left of the true-bearing line, the signal splitter sends an appropriate right or left signal to the BDI indicator. Thus, the true-bearing line serves as a reference axis for the BDI circuits. The true-bearing line is used in a similar way for simulating RLI circuits.

The vane in front of the photoelectric detector is used with the true-bearing line in simulating listening-pattern directivity. When the axis of the

SIMULATING RLI AND BDI

A real ship has BDI circuits and a real submarine has RLI circuits for obtaining accurate bearings. These circuits make use of the directivity of split transducers and hydrophones. Because the attack teacher does not have split hydrophones or transducers, it is necessary to simulate them. They are simulated by use of a position keeper operated by the optical-projector operator.

The position keeper controls the position of the true-bearing line, which is projected onto the screen. The submarine, or left-hand projector, system has an element that projects the line image onto the screen. This line always begins at the submarine because of the common mirror system. The line is approximately 2,500 scale-yards long, and it is radial to the center of the submarine. It can be rotated through 360° and is graduated in 500-yard steps, with range marks at 500, 1,000, 1,500, and 2,000 yards. The image-forming reticle is rotated by a synchro motor, which receives its orders from a synchro transmitter that is manually rotated by the operator. The projector operator's task is to manipulate the handwheel on the transmitter so that the beam of light at all times points directly to the pivoting point or center of the ship image on the screen. A synchro system is thus available in the equipment whereby simulated listening pattern is not exactly on the true-bearing line, the vane moves and the photoelectric detector develops a signal. This signal is used to vary the gain of an amplifier so that when the axis is off the beam, the intensity of the audio output is reduced.

Figure 17-5 shows the schematic diagram of the receiver-amplifier and BDI circuits used in the ship station. The target true-bearing repeater, B805, shown at the upper right corner of figure 17-5, receives its signals from the true-bearing-line transmitter at the optical projector. Its output drives one side of a mechanical differential. The other side of the differential is driven by the output of another differential, which is driven by both a ship's-compass repeater and a sound-beam relative-bearing repeater. Thus, the output of the second (upper) differential is the true bearing of the sound beam, and the output of the first (lower) differential is the bearing-angle off-train, as represented by the true-bearing line. The angle off-train output is coupled mechanically to the rotor of the signal splitter.

The stator of the signal splitter is energized by the echo signal. The magnitude of the two outputs of the rotor of the signal splitter depends on the angular position of the rotor with respect to the stators. This position in turn depends on the

FOLDOUT - Figure 17-5 -Schematic diagram of the receiver-amplifier and bear-ring-deviation indicator of the QFA-6.

angle off-train of the sound beam. The outputs are amplified in a twin-channel amplifier and then applied to the *comparison rectifier*, V831. The two diodes of the rectifier can conduct simultaneously, but the polarity of the output of the comparison rectifier depends on which diode receives the bigger signal. The d-c output of the comparison rectifier is amplified in a d-c amplifier and is applied to the horizontal-deflection coils of an oscilloscope indicator. The bearing-deviation indication is simply right or left motion of the oscilloscope spot.

The RLI circuits for the submarine station function like the BDI circuits just described for the ship station. When the hydrophone is trained so that its beam axis is not exactly on the true-bearing line, the angle off-train of the simulated submarine hydrophone is used to position a signal splitter similar to the one just described. The stator of the signal splitter is energized by the signal that simulates ship-propeller sounds. The outputs of the phase splitter are rectified by a twin diode and are used to energize a meter movement.

The projector operator who manipulates the true-bearing line thus functions as a position keeper-that is, he is responsible for indicating at all times the relative position of the two vessels. The angle off-train of the simulated sound beam (or the simulated hydrophone) from the true-bearing line energizes the BDI and RM circuits.

SIMULATING ACOUSTICS

Forming the Beam Pattern

In a real sonar, the operator sometimes uses the transducer as a hydrophone to listen for sound emitted from the target. The transducer beam pattern gives him directivity in his listening. To simulate this situation in the attack teacher, a special device is needed to sharpen the pattern for speed rotating carbon disk to make noises like a propeller. The output of the propeller-noise modulator is fed to a range attenuator, which governs the amplitude of the propeller noise as a function of range to the target. The output of the range attenuator is fed to the grid of the audio amplifier, V842, which is used as a gate tube. The amplification of V842 depends on a bias developed by the photoelectric cell, V858. The vane in front of the photocell is positioned by angle off-train. As the vane moves, it causes light from the d-c-excited light to fall onto the photoelectric cell. The voltage developed by the cell is applied as a bias to the grid of one section of V843, which is used as a d-c amplifier. The d-c amplifier is operated with the plate near ground potential and the cathode and grid returned to the o275 volt line. The output of the d-c amplifier is used to control the bias and gain of V842. Thus, as the sonar operator trains the transducer axis away from the true-bearing line, the vane controls the gain of the audio amplifier and produces a directional-pattern effect. The shape of the vane affects the "beam pattern."

Submarine Listening Equipment

The submarine sonar stack has a pattern-simulating circuit similar to that in the ship sonar console. The ship's propeller sounds, which are simulated in the submarine stack, are fed to an audio amplifier, the gain of which is controlled by a vane in front of a photoelectric cell. The position of the vane is controlled by the angle off-train of the listening hydrophone. Thus, as the operator trains his "hydrophone" off the true-bearing line, the audio output of the amplifier is reduced.

Echo Frequency

The echo frequency is determined by interrupting the d-c excited light beam that produces the active area. In the attack teacher this frequency is 800 cycles per second ± doppler shift.

listening, because the BDI simulator does not use a real transducer. The pattern is sharpened by a photoelectric cell, V858, and a vane attached to the differential output of angle off- train (figure 17-5), as referred to the bearing of the true-bearing line.

The submarine propeller sounds are simulated in the submarine propeller-noise modulator in the ship sonar console. The modulator uses a variable-

The beam is interrupted by a chopper disk attached to motor B206 (figure 17-3), which is a split-phase induction motor. This motor is driven by the motor amplifier. The signal input to the amplifier originates in another photoelectric detector, V203 and V204. The light falling on the photocells is interrupted by a disk attached to the echo motor, B221, which is a polyphase watt-hour meter motor. The speed of this motor is controlled

---

by the following factors: (1) The basic 800-cps frequency, (2) the speed and heading of the ship, (3) the speed and heading of the submarine, and (4) the relative bearing of the sound beam. The first factor is the basic audio frequency if no doppler is present. The second and third factors control the doppler shift that must be imposed on the echo frequency. The second and fourth factors control the doppler shift that must be imposed on reverberation echoes.

The basic 800-cps frequency is set by the *beat-frequency-oscillator (BFO) control*. This control is simply a variac which simulates the function of the BFO in a real sonar. The variac, T816, is shown at the lower left corner of figure 17-3. It controls the current through one of the current coils of the echo motor B221. The currents through the other current coils come from (1) the ship's coordinate-motor current coils, B213 and B214, and (2) the submarine's coordinate-motor current coils, B215 and B216.

The potential coils of B221 are energized from the phase-shifters, B806 and B203, which establish the relative motion of the two ships along the bearing line joining them. All these inputs combine to make the watt-hour meter motor, B221, rotate at the proper speed so that the light beam that falls on V203 and V204 is interrupted at

motor, B220, shown in figure 17-3 functions like the echo meter motor, B221, described previously. B220 has a disk, which interrupts the light that falls on photocell V202. The speed of B221 depends on ship speed and the relative bearing of the sound beam. Thus, the output of V202 is the audio frequency of own ship's doppler, which is modulated upon the reverberation noises developed in the ship sound-effect circuits. The ship sound-effect circuits also develop noises to simulate water noise and transmission-signal noise.

Range Attenuator

The intensity of the sounds heard at both ship and submarine should change as the range between ship and submarine changes. The attack teacher has a range attenuator that performs this function.

A control box on the projector contains a transmitter that rotates the true-bearing line. This control box contains a potentiometer with a dial calibrated in yards of range. The operator positions the potentiometer by estimating the range of the ship image from the submarine image. He uses the graduations on the true-bearing line to estimate the range. This rough estimate is sufficient for attenuation.

The potentiometer applies potential to two diodes,

800 cps ± doppler.

The motor drives a disk, which chops the light that falls on photoelectric cells V203 and V204. The output of the photoelectric cells is amplified in the two-phase push-pull motor amplifier and is used to drive the chopper-disk motor, B206. This motor is specially designed to follow rapidly all changes in excitation frequency. It carries the chopper, which interrupts the beam that develops the active area. Thus, the frequency of the modulation imposed on the beam is 800 cps ± doppler.

Reverberation

The attack teacher simulates the reverberation echoes heard immediately after each transmission. Own ship's doppler, which depends on own ship's speed and relative train of the sound beam, must be imposed on the reverberation echoes. The reverberations are developed in the *ship sound-effect* circuits, which are not illustrated schematically. These noises are developed by a thyraton and an 8.5-kc oscillator. The reverberation meter one in the acoustic amplifier of the submarine stack and the other in the acoustic amplifier of the sonar console. These diodes control the screen potentials, and hence the gain, of a stage in the acoustic amplifiers. Thus, as range changes, both the intensity of the ship-propeller sounds heard at the submarine stack and the intensity of the submarine-propeller sounds heard at the sonar console change.

Magnitude of the Doppler Effect

In the ocean the frequency of the sonar echo is affected by the motion of both the ship and the target because of Doppler effect. The frequency of the "echo" in the QFA attack teacher is the modulation of a projected light beam, which is interrupted by a chopper. The frequency of the modulation of the beam is varied in accordance with the motion of both ship and target by varying the speed of the motor that rotates the chopper. This motor is controlled by own ship's motion, target motion, and sound-beam bearing.

First consider the magnitude of the Doppler effect on a real ocean. The exact expression for the echo frequency is:

$$F_E = F_o(1 + 2V_1/v \cos \theta - 2V_2/v \cos \alpha),$$

where F_E is the echo frequency, F_o the transmission frequency, v the velocity of sound in water, V_1 the speed of own ship, V_2 the speed of the submarine, θ the relative bearing of the transducer on own ship, and a the angular difference between the submarine heading and true sound bearing.

Maximum doppler effect equals $2F_o/v$ times the relative velocity of the vessels. Assume that F_o, is 20,000 cycles per second and that v is 1,600 yards per second. If the velocity of the vessel is expressed in knots, v also must be expressed in knots. Because 1 knot is 2,000 yards per hour, it is 2,000/3,600, or 0.555, yard per second. Therefore,

$$2F_o/v = 2(20{,}000)(0.555) / 1{,}600 = 13.9.$$

The expression for audio echo frequency then becomes

When the echo is heterodyned in the receiver with a beat-frequency oscillator of frequency F_H, the output audio frequency, f_E, is

$$f_E = F_E - F_H$$

$$f_E = F_o - F_H + (2F_oV_1\cos\theta)/v - (2F_oV_2\cos\alpha)/v.$$

The expression "F_o-F_H" is the audio frequency of the echo if neither ship nor target is moving. It is also the reverberation frequency if the ship is not moving. If the term "F_o-F_H" is represented by f_o then

$$f_E = f_o + (2F_oV_1\cos\theta)/v - (2F_oV_2\cos\alpha)/v$$

Note that the second term represents the ship's motion. Therefore, the term,

$$(2F_oV_1 \cos\theta) / v,$$

can be defined as own ship's doppler (OD). Similarly, the third term,

$$(2F_oV_2 \cos\alpha) / v,$$

represents the effect of target motion only and is called *target doppler*, or TD.

Therefore, the audio echo frequency after heterodyning is

$$f_E = f_o + \text{OD} - \text{TD}.$$

The reverberation returns depend only on own ship's motion, and the audio frequency of the reverberation returns is given by the first two

$F_E = f_o + 13.9\ (V_1 \cos\theta - V_2 \cos\alpha)$ cycles per second.

With present-day speeds of ships, the doppler shift can change an 800-cps basic tone as much as 600 cycles per second-that is, from 200 to 1,400 cycles per second. In the attack teacher, therefore, the meter motors are designed to change the modulation of the sound beam and the frequency of reverberation noises between the limits of 200 and 1,400 cycles per second.

Doppler Nullifiers

In real equipment, nullifier circuits are added to the listening channels to compensate for own ship's doppler and target doppler. The own ship's doppler nullifier in a real sonar uses information from both own ship's speed and transducer heading in order to change the frequency of the beat-frequency oscillator and return the audio output to 800 cycles per second. Similarly, the target-doppler nullifier in a real sonar samples the first few cycles of the echo in order to correct the frequency of the beat-frequency oscillator to 800 cycles per second. Doppler nullifiers are not provided in the attack teacher, although nullifying can be done simply by not imposing ship's motion on the reverberation meter motor that establishes the basic 800-cps modulation.

Submarine Sonar Stack

The submarine sonar stack (figure 17-1) is-simpler than the ship sonar console. It includes an indicating range recorder, a remote training and bearing unit, and a receiver-amplifier with a separate loudspeaker.

The attack teacher projects the echo-ranging sound beam from the ship only. Thus, certain

terms in the previous equation-that is,

f_o+OD.

In the attack teacher, the frequency f_o is developed by chopping the light beam. The reverberation frequency is obtained by adding OD to f_o. The simulated echo frequency is obtained by adding OD and TD to f_o.

313

---

assumptions must be made in order to supply range information to the submarine without duplicating the echo-ranging devices. These assumptions are (1) that the submarine does not attempt to obtain the range of the ship until it has definitely established the ship's bearing by means of its listening equipment, and (2) that echo ranging is not continuous but is limited to the emission of infrequent single signals.

If a submarine and a ship were echo ranging simultaneously on each other, both vessels would hear (1) a transmission signal, (2) reverberation, (3) water noise, and finally (4) an echo. The time interval between the transmission and the echo would be the same for each vessel, but the frequencies of all the returns would be different because of the different doppler changes. The attack teacher uses the returns received at the ship sonar console in the submarine sonar stack to keep from duplicating the echo-ranging facilities. These returns give correct range but incorrect acoustic frequencies at the submarine station.

To include the single-ping feature in the submarine station, the input of the submarine sonar receiver of the attack teacher may be from one of two sources and is selected by a two-position switch. The switch positions are marked "echo range" and "listen." In the echo-range

Once the keying lock-out thyratron fires, any further keying pulse cannot actuate the keying circuits until the thyratron is deionized by changing the selector switch from *echo range to listen*.

At the end of the keying cycle-that is, when fly-back occurs-the input stage of the receiver becomes insensitive again and remains so until (1) the keying circuits are recycled by the selector switch, and (2) a new keying pulse again initiates a keying cycle. This arrangement provides the submarine sonar stack with echo-ranging facilities without duplicating the echo-ranging facilities of the ship.

No-Doppler Target

For some training operations, it is desirable to have a means of injecting a no-doppler target-that is, a target with no Doppler shift in frequency. Such a target can be used to train sound operators in identifying actual targets by the presence or absence of Doppler effect. It can be used also to produce an approximation of a wake echo. An accessory projecting device is mounted at the left end of the optical projector. It consists of (1) an image projector, which provides a red circular image on the screen, and (2) a telescope with a large objective, which is trained upon this image.

position the input of the submarine receiver is paralleled with the input of the ship receiver, and the input stage of the submarine receiver is made insensitive until the equipment is keyed. The keying pulse originates in the keying-control circuit of the ship and is the same pulse that initiates the cycle of events for the echo-ranging synthesis of the ship. Thus, both ship and submarine circuits are keyed simultaneously. The action of the keying pulse in submarine circuits is (1) to increase to normal the sensitivity of the input stage of the receiver, (2) to start the stylus drive circuit of the indicating-range recorder, and (3) to fire a keying lock-out thyratron. Therefore, as long as the ship has sound contact, the echo from this transmission is available also to the submarine equipment, which prints on the indicating-range recorder at the proper range. However, the frequencies of the reverberations and echo are incorrect, and the possibility of obtaining an echo has nothing to do with the bearing of the submarine hydrophone.

Adjustments are available for positioning this image to any portion of the ocean screen. The combination is similar to the arrangement of the submarine projector except that in the accessory projecting device, the image is not motor-driven. A photocell is located beyond the orifice of the telescope and is coupled to an amplifier. The electric signal produced when the sonar sound-beam traverses this image has a frequency that is correct for an echo from the attack-teacher submarine but that is incorrect for an echo from a no-doppler target. Therefore, this signal is rectified and used to key the *reverberation oscillator*, the output of which is used as the no-doppler target echo. The reverberation oscillator is keyed for an interval of time equal to the time for the sound beam to sweep across the no-doppler target. The no-doppler target therefore (1) has its own correct range and bearing indicated and (2) has a frequency that includes own ship's doppler but not target doppler.

AN/UQS-TI Sonar Training Set

DESCRIPTION

The AN/UQS-T1 sonar training set, or trainer, is a sonar problem generator that furnishes two or more synthetic sonar targets, in the form of artificial echoes, to a standard pulse-type scanning-sonar equipment. The synthetic targets are independently maneuverable in three dimensions, and the ship input to the trainer may be either actual own ship's motion or synthetic own ship's motion. Synthetic-target information is provided to all elements of the antisubmarine installation except the target depth-determining equipment. Control circuits are available for attaching a target depth-determining equipment trainer in the future. The equipment provides realistic training for all

switch on the front of the cabinet determines the mode of operation, as follows:

1. *Off*. When the switch is in this position all units of the equipment are turned off. This switch therefore acts as the main power switch for the entire equipment.

2. *Generate*. With the switch in this position, the motion of the ship is synthesized from the engine-telegraph voltage order and the rudder-telegraph synchro orders. These speed and rudder-angle orders originate at an external source, such as a mocked-up steering stand. Appropriate speed delays and turning delays are introduced automatically. The delay rates

members of the antisubmarine attack team, whether the ship is in port, under way on a fixed course, or engaged in attack maneuvers. The problem generators are constructed with an accuracy sufficient for use in precise tactical evaluation.

For shore-based or tender installations, an optical projector is provided that is similar to the projector of the model QFA-6. It projects onto a screen the image of own ship. Motion of the image on the screen represents the movement of the ship in the ocean. A target image representing the motion of the target also is projected onto the screen. When two to four targets are used, the coordinate motors of the projector are switched between targets. A person can stand behind the screen and can manually trace the path of each target with grease pencils so that the tracks of targets and surface ship may be plotted.

As shown in figure 17-6, the basic equipment consists of four major units-an own ship simulator, two sonar target simulators, and a transducer simulator. For shore-based or tender installations, a fifth major unit, an optical projector, is supplied. For installations in which the equipment must perform with maximum accuracy, a voltage regulator is also available.

Own Ship Simulator

The own ship simulator, as its name implies, contains circuits that generate factors of own ship's performance. A four-position selector are adjustable so as to cover various types of antisubmarine vessels. This position is used for shore and tender installations, as well as for ships on a fixed course or in port.

3. *Follow*. With the switch in this position, the ship image of the trainer follows the motion of own ship, using as inputs the orders from ship's gyro and pit log. This operating position may be used when it is desired to maneuver an anti-submarine vessel in a simulated attack.

4. *Calibrate*. This position is used for testing and calibrating the equipment. With the switch in this position the ship responds to the direct speed and rudder controls on the front of the cabinet.

Sonar Target Simulator

The two sonar target simulators contain all the circuits that generate factors of target performance. A target may be operated with a maximum speed of 30 knots, or, if a pair of gears in the calculating system is reversed, the target may have a maximum speed of 60 knots. Dials pertaining to speed are labeled appropriately to indicate the speed for which a target is set up. A submarine target should be arranged to have a maximum speed of 30 knots. Because accuracy of target motion is directly proportional to the speed of the motor drives, it is desirable to operate the drive motors at their maximum speeds for any given rate of target motion. Therefore, when the target is operated at a 30-knot maximum the 2-to-1

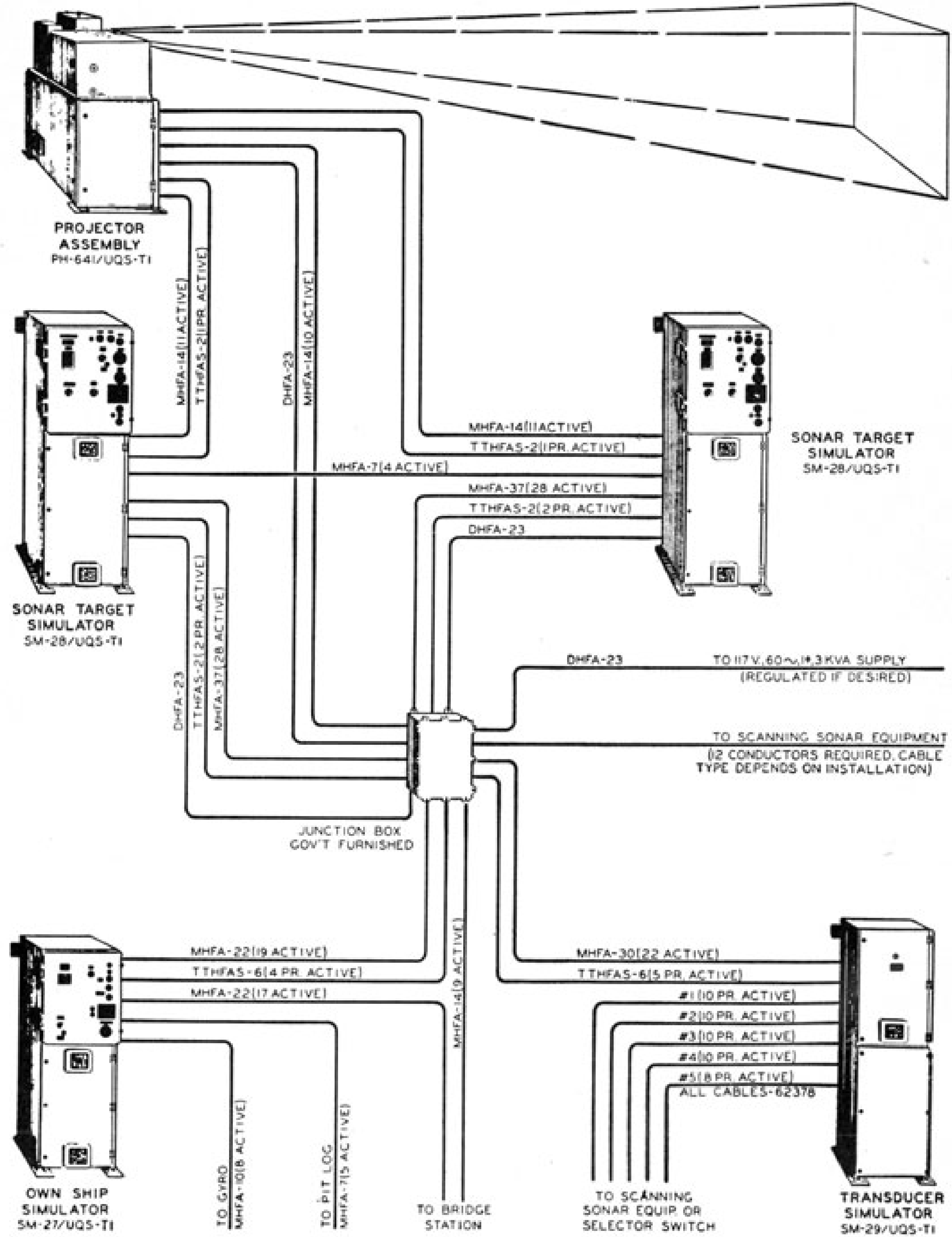


Figure 17-6 -Pictorial diagram of the AN/UQS-T1 equipment.

step-down of the gears ensures twice the accuracy of target motion in the lower speed range. In addition to the two speed ranges, each target operates in one of five modes selected by a single rotary switch, as follows:

1. *Normal*. In this mode the target functions as a normal submarine with all appropriate delays in acceleration and response to the helm. By means of adjustments, the tactical characteristics of any type of submarine can be duplicated.

2. *Reset*. In this mode the target may be positioned very rapidly (in 2 or 3 seconds) to any desired range and bearing, which are selected by two dials on the equipment. The maximum range of the target is 4,000 yards, and the maximum depth is 1,500 feet.

3. *Slave*. In this mode one target assumes the exact range and bearing of the other target but is incapable of producing echoes. This mode is required as a preliminary to the use of the second target in either of the two subsequent modes.

4. *Stop*. In this mode the second target remains fixed in the ocean and produces no-doppler echoes. This feature may be employed as a device for simulating an air bubble or knuckle by switching from *slave to stop*. Furthermore, it may be employed as a navigational aid, such as a sea buoy.

5. *Torpedo*. In this mode the second target functions as a submarine but without delays in acceleration or response to the helm. This mode may be used in simulating the firing of a torpedo by the controlling target because the torpedo must originate from the exact position in the ocean occupied by the firing vessel. The torpedo feature include reverberation, water noise, and ship's-screw noise in addition to the target-echo signal. There are no external controls in this unit. The signal outputs of the transducer simulator are 26-kc signals, which are sent to the receiver of the scanning-sonar equipment aboard the ship. These signals are coupled to the sonar receiving system through the scanning switches.

Optical Projector

The *optical projector* unit projects on a screen the light images representing own ship and target. A selector switch on the control panel at the rear of the unit (1) allows the selection of any one of a maximum of four targets or (2) provides for automatic sequencing of a maximum of four targets. Three automatic sequencing speeds are available. Indicator lights above the sequence selector switch indicate the operating mode of each target.

PRINCIPLES OF OPERATION

The trainer has two primary functions, as follows: (1) The production and indication of ship and target motion, and (2) the synthesis of acoustic information consistent with the conditions of the sonar problem. For installations employing the optical projector, the trainer has a third function-that of presenting the proper visual indication of the problem. In the following paragraphs these functions are discussed on a functional basis rather than by units.

A simplified functional diagram of the AN/UQS-T1 sonar training set is shown in figure 17-7. In the diagram rigid accuracy of connections has been sacrificed for simplicity. When "block numbers" are mentioned in the text, they refer to numbered units of figure 17- 7. Only one target is shown for simplicity.

can be used to represent the firing of a torpedo by the antisubmarine ship. This feature can be accomplished by keeping the target at zero range by means of the reset position until the time to fire.

Transducer Simulator

The *transducer simulator unit* accepts information generated by the own ship's simulator and the sonar-target simulators and modifies and converts this information into signals such as those that would be produced by a 48-element 19-inch magnetostriction transducer under the conditions of the sonar problem. These transducer signals

Ship Motion

When the own ship simulator unit is in the follow position the "trainer ship" follows the maneuvers of own ship. Synchro orders from the gyro-compass and the pitometer log cause mechanical rotations within the simulator that are representative of own ship's course and speed. When the selector switch is in the *generate* position the trainer ship is controlled by synthetic engine-telegraph, 1, and rudder-telegraph, 2, orders from

mocked-up ship controls. These orders cause mechanical rotations within the simulator with suitable acceleration and turning delays introduced. Adjustable delays provide for duplication of the tactical characteristics of the vessel to be simulated.

The mechanical system positioned in accordance with speed, drives a potentiometer in block number 9 that governs the output level of a power amplifier. The output voltage of this amplifier is proportional to speed. The mechanical system of ship's course positions a resolver, the rotor of which is excited by the voltage of ship's speed. The cosine and sine voltages from the stator winding s of a resolver thus represent N-S and E-W components of the ship's velocity.

These velocity signals constitute the inputs to two rate-servo mechanisms, which produce a speed of rotation that is proportional to the magnitude of the input voltage. The resultant motion of the N-S and E-W mechanical systems represents the resultant electric signal output of the synchro is the *relative* motion of ship and target in the N-S direction. In a similar manner, the relative motion of the target and own ship in the E-W direction is obtained as a synchro order.

Bearing Determination

The synchro orders representing E-W and N-S components of relative motion drive two mechanical systems. Each system drives the arm of a precision potentiometer, in block 18, that is excited by a fixed a-c voltage. The signal from the arm of the potentiometer to the midtap of the exciting transformer is defined as the component of horizontal range to the target, N-S in one system and E-W in the other system. The instantaneous polarity of the signal determines whether the range component is N-S or E-W. These two horizontal-range component signals are amplified by power amplifiers also in block 18.

The two-phase outputs of the power amplifiers are

components of ship's velocity in these directions. These mechanical systems thus follow the N-S and E-W Motion of the ship.

Each mechanical system drives a synchro transmitter at a constant rate of 200 yards per revolution, thus making available for external equipment the components of the movement of the ship in rectangular coordinates. These systems also drive suitable contact devices for the step motors of the attack plotter and the dead-reckoning tracer, thus replacing the Arma analyzer, which ordinarily drives this equipment.

In a shore-based projector assembly the ship's motion synchros directly govern the rotation of a pair of coordinate mirrors, which cause the image of the ship to move across the screen.

Target Motion

Each target unit contains controls for causing mechanical displacement in the target speed and course systems exactly the same as in the ship. If the aforementioned resolver methods are used, the mechanical outputs of two rate-servo mechanisms, 12, are the components of motion of the target in the N-S and E-W directions. Each mechanical integrator of target motion drives a 1DG differential synchro transmitter, DG_1, and DG_2 in figure 17-7. The north-south DG is excited by the N-S ship's motion synchro transmitter. The connected to the stator of a standard 5CT control transformer, CT_2. The range signals have identical a-c *time* phase but may be considered to constitute a two-*space*-phase system. The conventional synchro order constitutes a three-*space*-phase system. One system may be converted to the other by precisely the same electric connections that are required for conversion from two-*time*-phase system to a three-*time*-phase system. The rotor signal of the CT_2 excites a wipe-out servo-amplifier system, 21. The rotor angle of the CT_2 at servo balance is an angle the tangent of which is the ratio of the E-W voltage to the N-S voltage. This angle is, by definition and calibration, the bearing of the target, *Br*.

Various synchros and resolvers are also positioned by the bearing-solver mechanism. The bearing is transmitted at 1 and 36 speed to provide target-bearing information for use in the projector assembly and the transducer simulator. These bearing transmitters are designated G_1 in figure 17-7.

Horizontal-Range Determination

A second 5CT, CT_1, is driven by the bearing-solver mechanical system just described. The stator is connected in parallel with the stator of CT_2. However, its rotor is physically displaced 90°. Thus, when the rotor voltage of the bearing-solver, CT_2, is zero the rotor voltage of the

FOLDOUT - Figure 17-7 -Simplified functional diagram of AN/UQS-T1.

horizontal-range synchro, CT_1, is a maximum.

The value of these rotor voltages is a function of the magnitudes of the E-W and N-S horizontal-range voltages. The result of the special relation of the stator windings is that the rotor voltage is proportional to the square root of the sum of the squares of the range-component voltages. This signal is the horizontal range of the target.

Target Depression-Angle Solution

The horizontal-range voltage is amplified in the Rh amplifier, 17, and connected to one set of coils of a 5CT control transformer, CT_4. An adjustable autotransformer, 15, on the panel of the target unit is calibrated, in feet, for target depth and also delivers a voltage to CT_4 that is proportional to the depth of the target, *Hq*. The horizontal-range and depth voltages are connected by a two-phase to three-phase connection, 16, similar to that employed in the range-component circuits. The CT_4 is driven mechanically by a servo system, 22, that responds to the rotor voltage of CT4. The result is that the system rotates to an angle the tangent of which is the ratio of depth to horizontal range. This angle is the true-depression angle *Et* of the target.

Slant-Range Determination

A second 5CT control transformer, CT_3, is connected in parallel with the depression-angle solver, CT_4, and its rotor is driven mechanically by the depression-angle servo system. In the same manner as the horizontal-range synchro, CT_1, the rotor of the slant-range, CT_3, is zeroed so that when the servomechanism has solved the depression angle, a signal appears at the rotor terminals of the slant-range synchro. This signal voltage is proportional to the square root of the 10 to 15 cycles per second. Therefore, the system must be extremely accurate because the acoustic synthesis is at the transducer frequency, which is approximately 26 kc. In addition, miscellaneous acoustic effects such as reverberation, propeller sounds, and water noise must be synthesized to provide a realistic trainer. The basic output signal of the trainer is a 26-kc signal varied in frequency by the frequency-control system.

Frequency-Control System

A master oscillator, 3, in own ship's simulator operates at a frequency of 24 kc and is mixed with the 26-kc output of a reactance-tube controlled oscillator. The beat frequency is the input to a discriminator that is tuned to a fixed frequency of 2 kc. Immediately after the equipment is keyed by the scanning-sonar, the reactance-tube control grid is connected momentarily to the output of this discriminator. This connection causes the reactance-tube controlled oscillator to change frequency until it reaches a frequency that is equal to the sum of the master-oscillator frequency and the frequency to which the discriminator is tuned. After this "sampling" the reactance-tube grid is disconnected from the discriminator, but a large capacitor maintains the same potential until the next sampling interval.

Target Echo-Frequency Control

The 24-kc master-oscillator frequency is delivered to each of the target units, where an identical arrangement assures that the local oscillator within each target attains the same frequency during the sampling period as was attained by the local oscillator in the own ship simulator. At the end of the sampling period a second reactance tube in the local oscillator of the target is biased by a voltage, the magnitude and polarity of which are proportional to the amount of target Doppler. This condition causes the frequency of the target-local

sum of the squares of horizontal range and depth. This signal, then, is the slant range, *Rq*. This voltage is compared with the voltage from a precision potentiometer excited by a fixed a-c signal. The difference in magnitude provides a signal to a wipe-out servo system that drives the arm of the potentiometer until its voltage equals the slant-range voltage. By calibration, the motion of this system is the slant range of the target.

Acoustic Synthesis

The primary problem of acoustic synthesis is the faithful reproduction of Doppler effect for each target. The Doppler effect must be correct within oscillator in block 20 to differ, in frequency, from that of the ship's oscillator, block 3, by the magnitude of the Doppler effect.

The target-Doppler effect is controlled by a resolver, in block 20, excited by target speed, the rotor being turned mechanically by the difference between target heading and true bearing (target angle). The resulting signal is a voltage that is proportional to the component of velocity of the target along the line of bearing. This signal is rectified to operate the doppler-reactance tube.

Echo Timing

In the own ship simulator unit a d-c voltage is generated in block 7. This voltage, starting at zero when the sound equipment keys, increases linearly to approximately 110 volts in 5 seconds. The voltage is delivered to each of the targets in the system, where it supplies the grid signal for a thyratron. The cathode of this thyratron is established at a d-c potential by a potentiometer driven by the slant-range mechanical system. The combination is such that the thyratron fires when the sweep voltage is approximately the value of the cathode voltage. By calibration, the thyratron fires at the precise time for an echo to return from a target. The slant range of the target is indicated by the system, if a sound velocity of 4,800 feet per second is assumed. This thyratron causes a trigger circuit to introduce a short pulse of the target-echo frequency. The length of the pulse is governed by the aspect of the target, which is determined by a resolver that compares the difference between target head and true bearing. For a beam aspect a 35-millisecond pulse is produced; for a stern or bow aspect the pulse is about three times longer, which represent, in magnitude and phase, the signals that would exist in a scanning transducer actuated by a plane-front sound wave. The purpose of the lag line is very similar to that in the QGB series described in chapter 6. These signals are connected to the segments of the rotor and therefore appear at the stator terminals representing the relative bearing of the target. The stator terminals are connected through 100-ohm resistors to ground, and the transducer cables of the scanning-sonar equipment are connected to the stator terminals. The 100-ohm resistors represent the transducer electrically. The additional scanning switches required by additional targets are connected in parallel on the stator side, the 60,000-ohm reactance of the capacitance of each segment constituting adequate decoupling between the various targets.

Reverberation Synthesis

The local oscillator, 3, in own ship simulator is the no-doppler frequency; hence, it is the reverberation frequency. A reactance tube in the local-oscillator circuit of the ship shows the true character of reverberation by giving a random fluttering signal to

and the power level of the signal is greatly diminished. Furthermore, the power level of the echo is attenuated automatically by the d-c slant-range voltage, which governs the firing time of the echo thyratron.

Production of Transducer Signals

The target-echo signal is delivered to the transducer simulator, where it is applied to the slip rings of a device that closely resembles the scanning switch or capacity commutator of the QHB-series scanning-sonar equipment. This device is given the name "scanning switch," but it is not identical with the scanning switch used in the QHB series. The rotor of the scanning switch in the training equipment is positioned to the relative bearing of the target by a servomechanism in response to 1-speed and 36-speed synchro orders from a pair of 1DG differential transmitters. The rotors of these 1DG's are driven by the true-bearing mechanical system of the target. The stator excitation is 1-speed and 36-speed gyrocompass orders. The output of the 1DG's is the relative bearing of the target from the ship. The rotor of the scanning switch is positioned to this angle.

The lag line (phase shifter) on the rotor converts the target-echo signal input into an array of signals, produce "wobble" of the reverberation frequency. When the sound equipment is keyed, this frequency is delivered to a circuit that provides for full output.

Following the initiation of the keying pulse, there is a gradual decay with respect to time. Both volume and duration may be adjusted by controls at the ship unit. The basic signal is delivered to a 48-segment ring line in the transducer simulator through a series of magnitude-wobbling circuits. Each segment of the ring line is connected through a small capacitance to the stator terminals of the echo-bearing switches. At the terminals of the echo-bearing switches is an array of signals representing, in duration and direction, typical reverberation patterns.

Propeller Sounds

An irregular-contact device, 19, in the ship unit is driven by a motor at a speed that is proportional to the speed of the ship and the magnitude of the output is controlled by potentiometer, P_2, excited by target speed through a servo amplifier, 14. The signals from this contact device modulate gas-tube noise. The output of the circuit is connected to appropriate points on the ring line of the

transducer simulator and causes the ship's propeller sound when the ship is operating at high speeds. In a similar manner, propeller sounds of the target are introduced at the target-bearing switch rotor so that the target propeller sounds appear at the proper bearing. The circuits are arranged so that the target sounds are missing for speeds below 5 knots.

Water Noise

At high ship speeds, omnidirectional random noise, or water noise from own ship, is introduced into the transducer-simulator output. This noise rapidly increases as the speed is increased by a servo system, 8, excited by ship's speed which controls a potentiometer, P_1, to vary the output magnitude in proportion to speed.

Dome Baffle

To depict the appearance and sound of a dome baffle, the reverberation ring-line connections are deleted at the after elements of the echo-bearing switch stators. Thus reverberation or ship's sounds are not audible or visible for several degrees about the stern. In a more complex manner the target-echo and propeller sounds are suppressed by synchro methods when the bearing of the target is within 20° of the stern. The baffle effect can be eliminated by a switch on the console.

MCC Operation

For maintenance-of-close-contact (MCC) operation, a pair of 1G synchro transmitters (not shown in figure 17-7) is driven at 2 speed and 36 speed by the depression-angle mechanism in each target. These transmitters (1) provide the basic information for a future trainer to be used with the target depth-determining equipment and (2) control the effect of lost contact due to target fact is an important reminder to sound operators that MCC is for close-range operation only.

Optical Projector

A rotatable reticle defines (1) the ship image, (2) the angular position of this reticle, and hence (3) the heading of the ship image on the screen. The reticle is controlled by the ship's-course synchro order, which originates in the ship's-course generator of own ship simulator.

Motion of the ship image on the screen is controlled by a pair of coordinate mirrors in a manner similar to that of the QFA-6 equipment.

Each target unit contains in the true-bearing mechanical system a pair of miniature synchro transmitters, GI, operating at 1 speed and 36 speed. If it is desired to depict a specific target, these transmitters are connected by a relay to a pair of miniature control transformers in a mechanism of the projector assembly. By servo action this mechanical system rotates to a position equal to the true bearing of the target. This system also rotates a turntable at one revolution for 360° of bearing. The center of the turntable is a tube, through which the rotatable image of the target is projected by a lens, an image reticle, and a light source. The light from this system is diverted at right angles by a prism at the outboard end of the tube to a rotatable mirror, which diverts the light back nearly parallel to the axis of the tube. The light then strikes the coordinate mirrors, which project the ship image on the screen. The angular position of the mirror diverting the light from the prism is controlled by a servo, which moves in accordance with the horizontal range of the target. The result is that the target image is positioned with respect to the ship image in accordance with the range and true bearing of the target, as governed by the target rangekeeper. Horizontal range is obtained mechanically by comparing the horizontal-range voltage of the target

depth.

When the target-depth angle exceeds 30°, the synchro system actuates a blocking circuit that causes the echo from the particular target to disappear. Relays connected to the MCC control line of the scanning-sonar equipment disable this blocking circuit but reduce the power level of the echo and of the reverberation. The echo strength is reduced so greatly that contact is difficult to maintain at ranges beyond 1,500 yards. This

with the voltage from the arm of a precision potentiometer, which is excited from a fixed voltage. The difference voltage drives a wipe-out servo system, which positions the potentiometer arm.

The result of this projection scheme is that the horizontal range and bearing of the target as they

---

appear on the screen must always be in agreement with the situation indicated at the sonar target simulator.

Additional Targets

For certain shore-based training, more than two targets may be required. The range-integration transmitters of the ship unit are of adequate size to allow the addition of any number of targets to the system. A transducer simulator must be added for each two targets, because the echo-bearing switches are a part of the transducer simulator and one switch is required for each target. An interesting detail of this system is that own ship's motion input to all targets is identical-

a fact that should be useful for accurate analysis of complex maneuvers.

Adaptation to Searchlight Sonar

If desired, the output of the transducer simulator may be converted to that of a searchlight transducer for training searchlight-sonar operators. A standard QHB audio scanning switch, connected to the transducer simulator, and positioned by a 1-speed and 36-speed relative-bearing synchro order from the searchlight equipment, produces the required signal to the equipment receiver. If a split transducer for bearing deviation indication is to be represented, a double-beam audio switch is required.

Previous Chapter Sonar Home Page Next Chapter

APPENDIX

DEFINITIONS OF FIRE CONTROL TERMS AND SYMBOLS

AS APPLIED TO SONAR

DUAL SINGLE-AXIS SYSTEM -This term refers to a system of gimbaling employed to position two planes in space each of which rotates about a single axis of rotation. In a typical sonar system one of the planes is a sound plane, perpendicular to the deck, used to determine the bearing of the sonar target. The other is a sound plane, parallel to the deck, used to determine the depression of the sonar target. A dual single-axis system is shown in figure 9-8.

LINE OF SIGHT -The line of sight as used in this text refers to a straight line from the point of origin to its termination.

LINE OF SOUND -The line of sound as used in this text refers to a line extending from the sound head along the sound-beam path (figure 9-10) or along the line of intersect ion of two single-axis sound planes (figure 9-12). It deviates from a straight line by the amount the sound beam is refracted by temperature, salinity, and pressure gradients present in the water medium through which the sound travels.

SONAR DECK PLANE -AS used in this text, the sonar deck plane is the plane through the center of the depth-determining-equipment transducer and parallel to the reference surface of the stable element.

SONAR HORIZONTAL PLANE -AS used in this text, the sonar horizontal plane lies on the horizontal plane through the center of the depth-determining-equipment transducer.

SOUND BEAM -This term is used to designate

Br-RELATIVE TARGET BEARING -The angle between the vertical plane through the fore-and-aft axis of own ship and the vertical plane through the line of sight from the main director, measured in a horizontal plane clockwise from the bow.

B'r-STABLE ELEMENT TRAIN -The angle between the fore-and-aft axis of own ship and the vertical plane through the line of sight from the main director to the target for which the stable element is trained, measured in the deck plane clockwise from the bow.

Brq-RELATIVE SONAR TARGET BEARING -The angle between the vertical plane through the fore-and-aft axis of own ship and the vertical plane through the line of sight to the sonar target, measured in a horizontal plane clockwise from the bow.

B'rq-STABILIZED SONAR TRAIN -The angle between the fore-and-aft axis of own ship and the vertical plane through the axis of the sound beam at the sound head measured in the deck plane clockwise from the bow. The stabilized sonar train is normally used in referring to three-axis systems.

B'r'q-AZIMUTH SONAR TRAIN -The angle between the fore-and-aft axis of own ship and the plane perpendicular to the deck through the line of sight to the sonar target. measured in the sonar deck plane clockwise from the bow.

Co-OWN SHIP'S COURSE -The angle between the north-and-south vertical plane and the vertical plane through the fore-and-aft axis of own ship, measured in a horizontal plane clockwise from the north. This angle is normally measured by the gyrocompass.

the pattern formed by lines of sound emitted from a sound head mounted on three mutually perpendicular axes.

SOUND HEAD -The term "sound head" is used to designate the sonar transducer or hydrophone including its housing and mounting.

SOUND PLANE -The term "sound plane" is used to designate the pattern formed by the sound emitted from a sound head mounted on a single axis of rotation.

THREE-AXIS SYSTEM -This term refers to a system of gimbaling employed to position a line in space about three axes of rotation relative to the deck of the ship. These axes are a train axis perpendicular to the deck, a crosslevel axis parallel to the deck, and a level axis, in the horizontal plane perpendicular to the crosslevel axis. The line positioned may be a line of sight, the axis of a sound beam, or any other line. A three-axis system is shown in figure 9-10.

TRANSDUCER -The transducer is the sensitive receiving and transmitting element of a sound head. * * * * Bq-TRUE SONAR TARGET BEARING -The angle between a north-and-south vertical plane and the vertical plane through the line of sight to the sonar target, measured in a horizontal plane clockwise from the north.

Eq-APPARENT DEPRESSION ANGLE -The angle of depression below the sonar horizontal plane of the acoustic path of the QDA transducer to the sonar target, measured in the vertical plane through the line of sight to the sonar target.

Eq differs from Etq by the effect of refraction because of variations in velocity in the water caused by temperature salinity, and pressure gradients. For "on target" indications

cEtq ± jEtq = Eq.

ΔEq-INCREMENT OF SONAR DEPRESSION -Changes in apparent depression angle (Eq).

E'q'-SONAR ACOUSTIC DEPRESSION FROM DECK -The angle between the direction, at the QDA transducer, of the acoustic path to the sonar target and the sonar deck plane, measured in a plane through the line of sight to the sonar target and perpendicular to the sonar deck plane.

Eqr-REFRACTED DEPRESSION ANQLE -The actual depression angle of the sound beam after passing through the thermocline.

E'q's-"ON TARGET" DEPRESSION OF DEPTH-DETERMINING-EQUIPMENT TRANSDUCER BEAM -The angle of the

beam of the depth-determining-equipment transducer from the sonar deck plane, measured in the plane through the fore-and-aft axis of own ship and perpendicular to the deck, when an "on target" indication is obtained. This angle is the tilt of the transducer.

Etq-SONAR TARGET DEPRESSION -The angle of depression below the sonar horizontal plane of the line of sight to the sonar target, measured in a vertical plane through the line of sight to the sonar target.

cEtq-COMPUTED SONAR TARGET DEPRESSION -The predicted sonar target depression (Etq), determined by the OKA-1 equipment from the velocity of sound in the water, the latest indications of the depth of the thermocline and sound range (Rq), and the target depth below transducer (H'q)i This computed angle is used to assist the depth-determining-equipment operator to follow the sonar target with the acoustic beam.

jEtq-ADJUSTMENT OF COMPUTED SONAR TARGET DEPRESSION -The adjustment of the computed sonar target depression (cEtq) by the depth-determining-equipment operator to correct for deviations of the acoustic beam off the target as indicated by the depth-deviation indicator.

Hq-SONAR TARGET DEPTH -The vertical depth of the sonar target below the surface of the water.

Hg-RELATIVE TARGET DEPTH -The depth of the target relative to the depth of the depth-determining-equipment transducer.

L-LEVEL ANGLE -Measured about an axis in the horizontal plane; it is the angle (figure 9-6) between the horizontal plane and the deck plane, measured in the vertical plane through the line of plane and the deck plane. The roll is positive when the starboard side of the ship is up. The angle of roll is shown in figure 9-6.

N-PITCH -Measured about a horizontal axis; it is the angle measured about the intersection of the horizontal plane with the athwartship plane perpendicular to the deck, between the vertical plane and a plane perpendicular to the deck through this axis. This pitch is positive when the bow of the ship is up. The angle of pitch is shown in figure 9-6.

PVq-DEPTH OF DDE TRANSDUCER -Depth of the transducer below the surface of the water.

Rq-SOUND RANGE -The distance from the center of the azimuth transducer to the sonar target, measured along the sound path.

Rhq-HORIZONTAL SOUND RANGE -The projection of the sound range (Rq) on a horizontal plane.

dRhq-HORIZONTAL SOUND RANGE RATE -The time rate of change of horizontal sound range (Rhq).

V or v-VELOCITY OF SOUND IN MIXED LAYER -The velocity of sound in the layer of water near the surface of the ocean.

ΔV-VELOCITY DECREASE THROUGH THE THERMOCLINE -The decrease in the velocity of sound in passing through the thermocline.

VZ-VERTICAL COMPONENT OF THE VELOCITY OF SOUND-The speed at which a sound beam travels downward.

Zd-CROSSLEVEL ANGLE -Measured about an axis in the deck; it is the angle (figure 9-6) that is measured about the intersection of the plane of the deck with the vertical plane through the line of sight

sight. This angle is positive when the deck toward the target is below the horizontal plane. The level angle is shown in figure 9-6.

Lq-SONAR LEVEL ANGLE -Measured about an axis in the horizontal plane; it is the angle between the horizontal plane and the deck plane measured in a vertical plane through the axis of the sound beam at the sound head. This angle is positive when the deck toward the target is below the horizontal plane. The sonar level angle is shown in figure 9-9.

M-ROLL -Measured about an axis in the deck; it is the angle measured in the athwartship plane perpendicular to the deck between its intersection with the horizontal

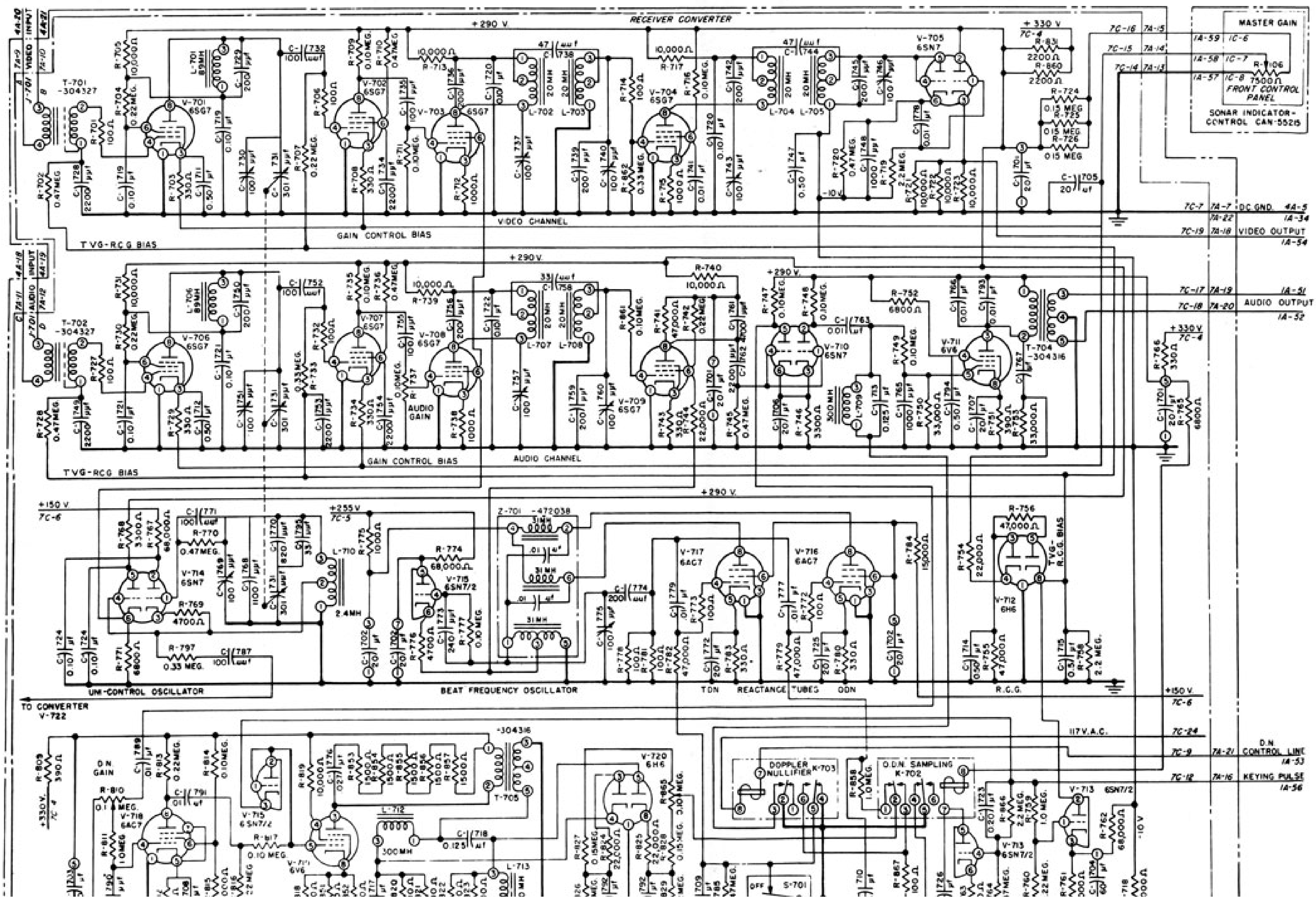
for which the stable element is trained, between the vertical plane and a plane perpendicular to the deck through this axis. This angle is positive if, when facing the target, the deck on the observer's right is up. The crosslevel angle is shown in figure 9-6.

Zdq-SONAR CROSSLEVEL ANGLE -Measured about an axis in the deck ; it is the angle which is measured about the intersection of the plane of the deck with the vertical plane through the axis of the sound beam at the sound head, between the vertical plane and a plane perpendicular to the deck through this axis. This angle is positive if, when facing the target, the deck on the observer's right is up.

INDEX

Figure 7-2. -Dual-channel receiver circuits.

Sonar Home Page



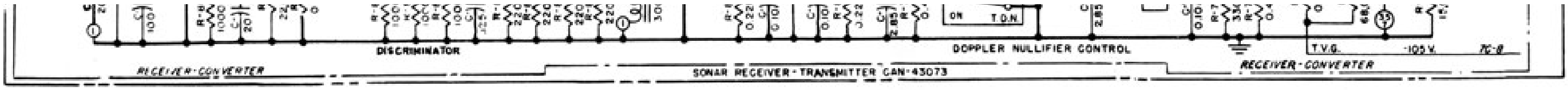
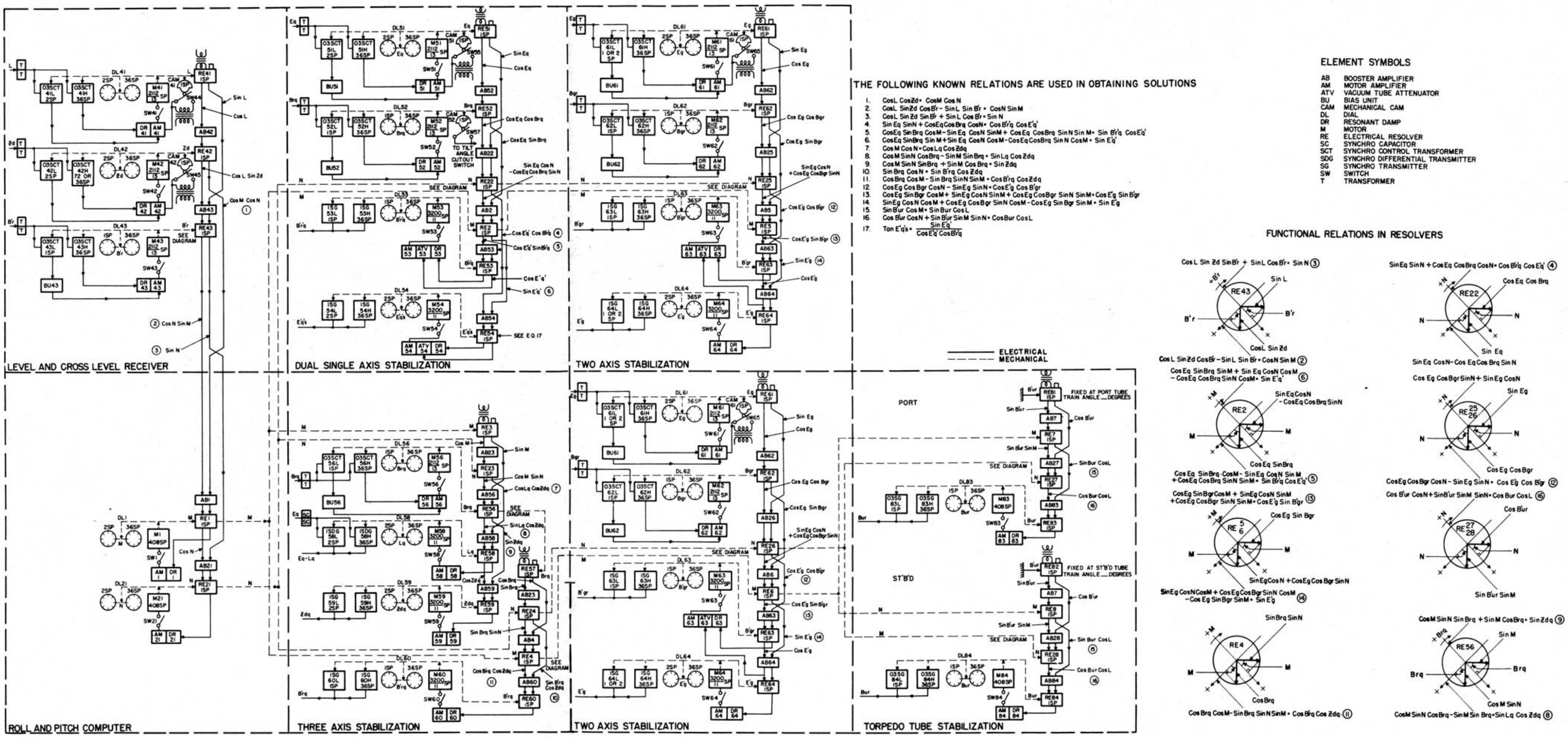

DISCRIMINATOR
DOPPLER NULLIFIER CONTROL
T.V.G.
-105 V.
RECEIVER-CONVERTER
SONAR RECEIVER-TRANSMITTER CAN-43073
RECEIVER-CONVERTER

Figure 9-11 -Functional diagram, Mk 59 Mod 0.

Sonar Home Page



THE FOLLOWING KNOWN RELATIONS ARE USED IN OBTAINING SOLUTIONS

1. CosL CosZd= CosM CosN
2. CosL SinZd CosB'r - Sin L Sin B'r = CosN SinM
3. CosL SinZd SinB'r + Sin L Cos B'r = Sin N
4. SinEq SinN + CosEqCosBrq CosN = Cos B'r'q Cos E'q'
5. CosEq Sin Brq CosM-Sin Eq CosN SinM + Cos Eq CosBrq Sin N Sin M= Sin B'r'q Cos E'q'
6. CosEq SinBrq Sin M +Sin Eq CosN CosM-CosEq CosBrq SinN CosM = Sin Eq'
7. CosM CosN = CosLq CosZdq
8. CosM SinN CosBrq- SinM SinBrq = SinLq CosZdq
9. CosM SinN SinBrq + SinM Cos Brq = SinZig
10. Sin Brq CosN = SinB'rq CosZdq
11. CosBrq CosM - SinBrq SinN SinM = CosB'rq CosZdq
12. CosEg CosBgr CosN - SinEg SinN = CosE'g CosB'gr
13. CosEg SinBgr CosM+ SinEg CosN SinM + CosEg CosBgr SinN SinM = CosE'g Sin B'gr
14. SinEg CosN CosM + CosEg CosBgr SinN CosM -CosEg SinBgr SinM = SinE'g
15. SinB'ur CosM = SinBur CosL
16. CosB'ur CosN +Sin B'ur SinM SinN = CosBur CosL
17. Tan E'q's = SinE'q' / Cos E'q' CosB'r'q

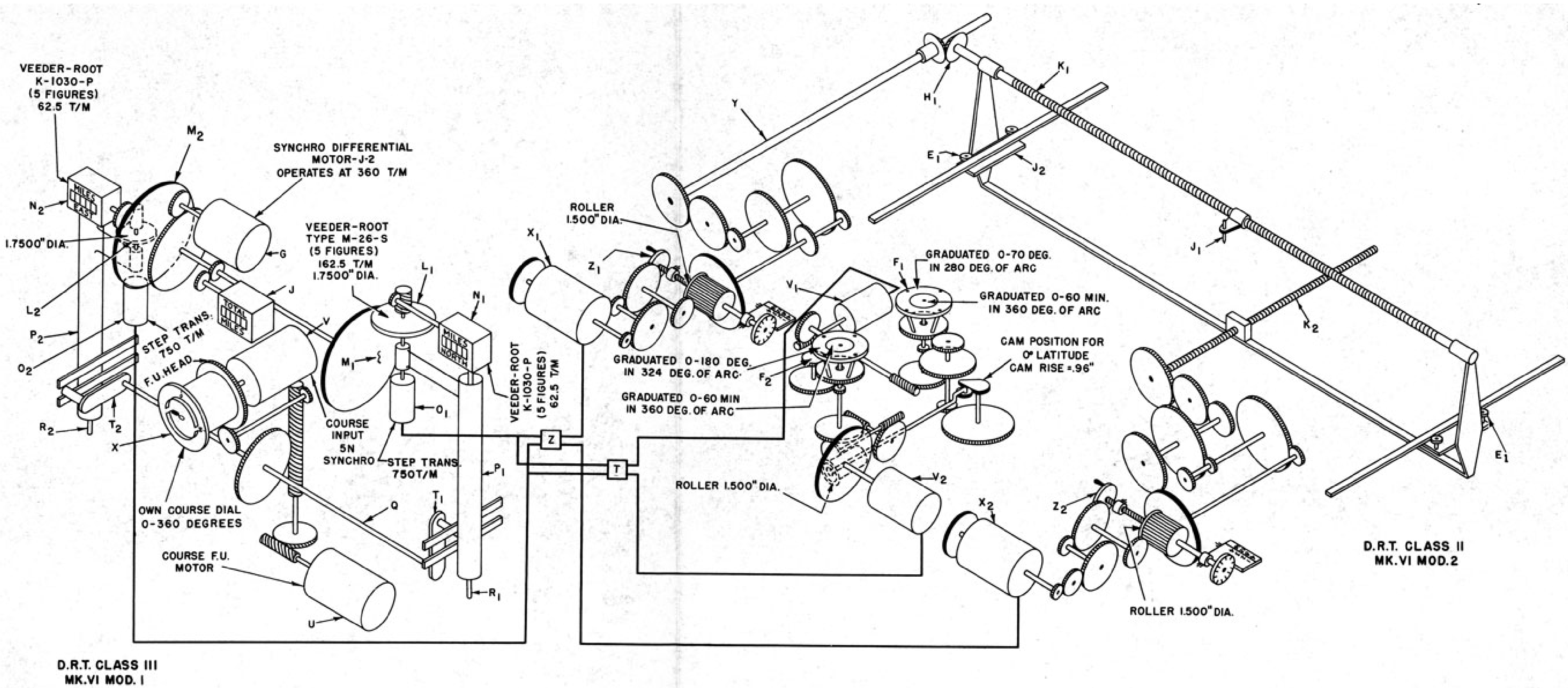
ELEMENT SYMBOLS

AB	BOOSTER AMPLIFIER
AM	MOTOR AMPLIFIER
ATV	VACUUM TUBE ATTENUATOR
BU	BIAS UNIT
CAM	MECHANICAL CAM
DL	DIAL
DR	RESONANT DAMP
M	MOTOR
RE	ELECTRICAL RESOLVER
SC	SYNCHRO CAPACITOR
SCT	SYNCHRO CONTROL TRANSFORMER
SDG	SYNCHRO DIFFERENTIAL TRANSMITTER
SG	SYNCHRO TRANSMITTER
SW	SWITCH
T	TRANSFORMER

239276 0-53 (Face p. 192)

Figure 11-3 -Dead-reckoning system.

Sonar Home Page



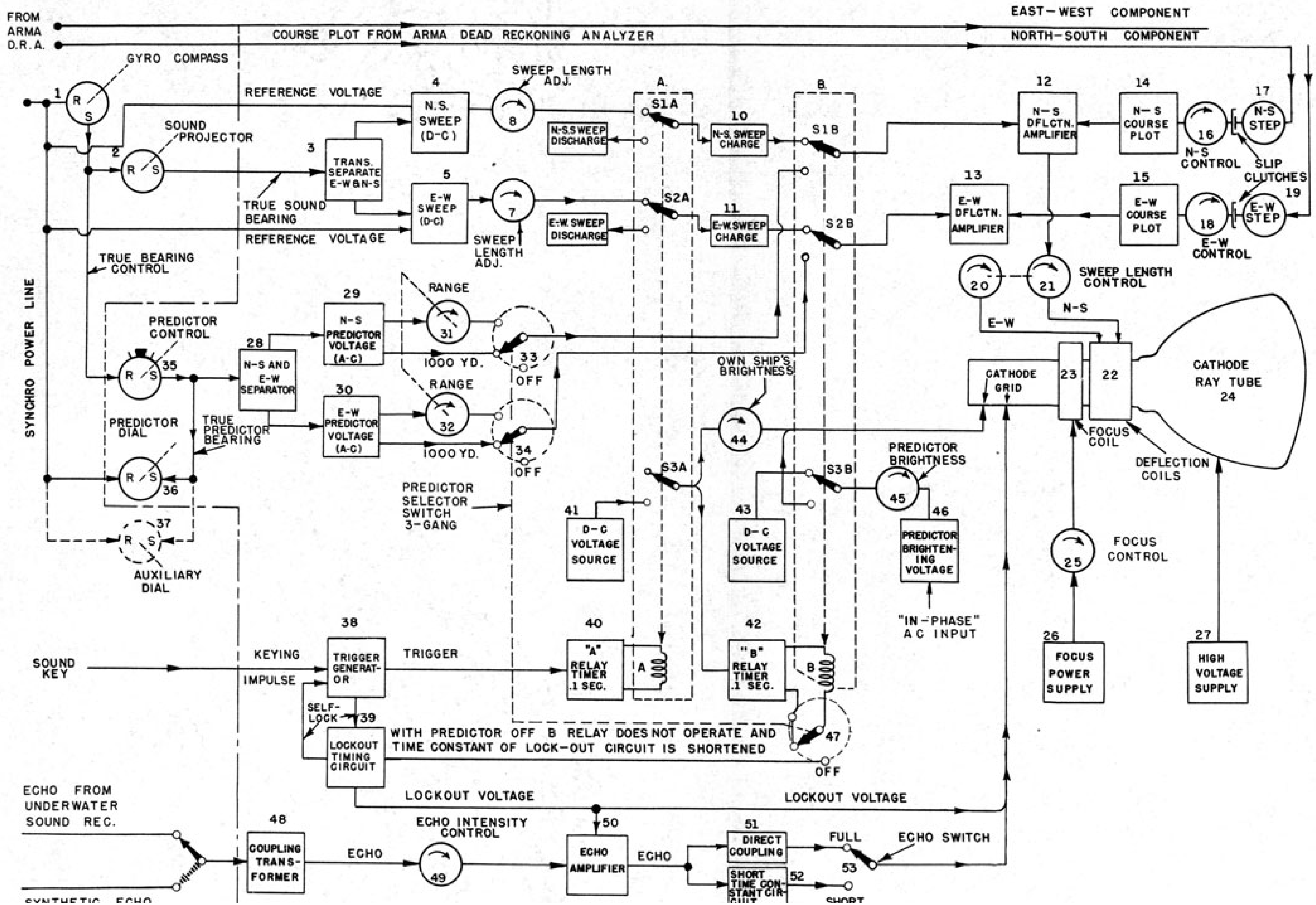
D.R.T. CLASS III, MK.VI MOD. I
D.R.T. CLASS II, MK VI MOD. 2

G-DISTANCE INPUT MOTOR	Y-FLUTED SHAFT	K_1-TRANSVERSE LEAD SCREW	R_1-ROLLER GUIDE	J_2-PENCIL CARRIAGE	P_2-EAST ROLLER CARRIAGE
J-TOTAL MILES COUNTER	Z-COMPONENT INTERCHANGE SWITCH	L_1-NORTH SPEED ROLLER	T_1-NORTH CRANK ARM	K_2-LONGITUDINAL LEAD SCREW	R_2-ROLLER GUIDE
Q-CRANK SHAFT		M_1-NORTH COMPONENT DISC	V_1-LATITUDE MOTOR	L_2-EAST SPEED ROLLER	T_2-EAST CRANK ARM
T-DIAL REVERSING SWITCH	E_1-ROLLER GUIDES	N_1-NORTH REVOLUTION COUNTER	X_1-CROSS SCREW MOTOR	M_2-EAST COMPONENT DISC	V_2-LONGITUDE MOTOR
U-COURSE FOLLOW-UP MOTOR	F_1-LATITUDE DIALS		Z_1-LATITUDE MILES-PER-INCH ADJUSTING MECHANISM	N_2-EAST REVOLUTION COUNTER	X_2-LEAD SCREW MOTOR
V-5N SYNCHRO MOTOR	H_1-BEVEL GEARS	O_1-NORTH STEP TRANSMITTER			Z_2-LONGITUDE MILES-PER-INCH ADJUSTING MECHANISM
X-COMPASS DIAL	J_1-PENCIL CARRIER	P_1-NORTH ROLLER CARRIAGE	F_2-LONGITUDE DIALS	O_2-EAST STEP TRANSMITTER	

239276 0-53 (Face p. 216)

Figure 11-8 -Complete block diagram of the attack plotter.

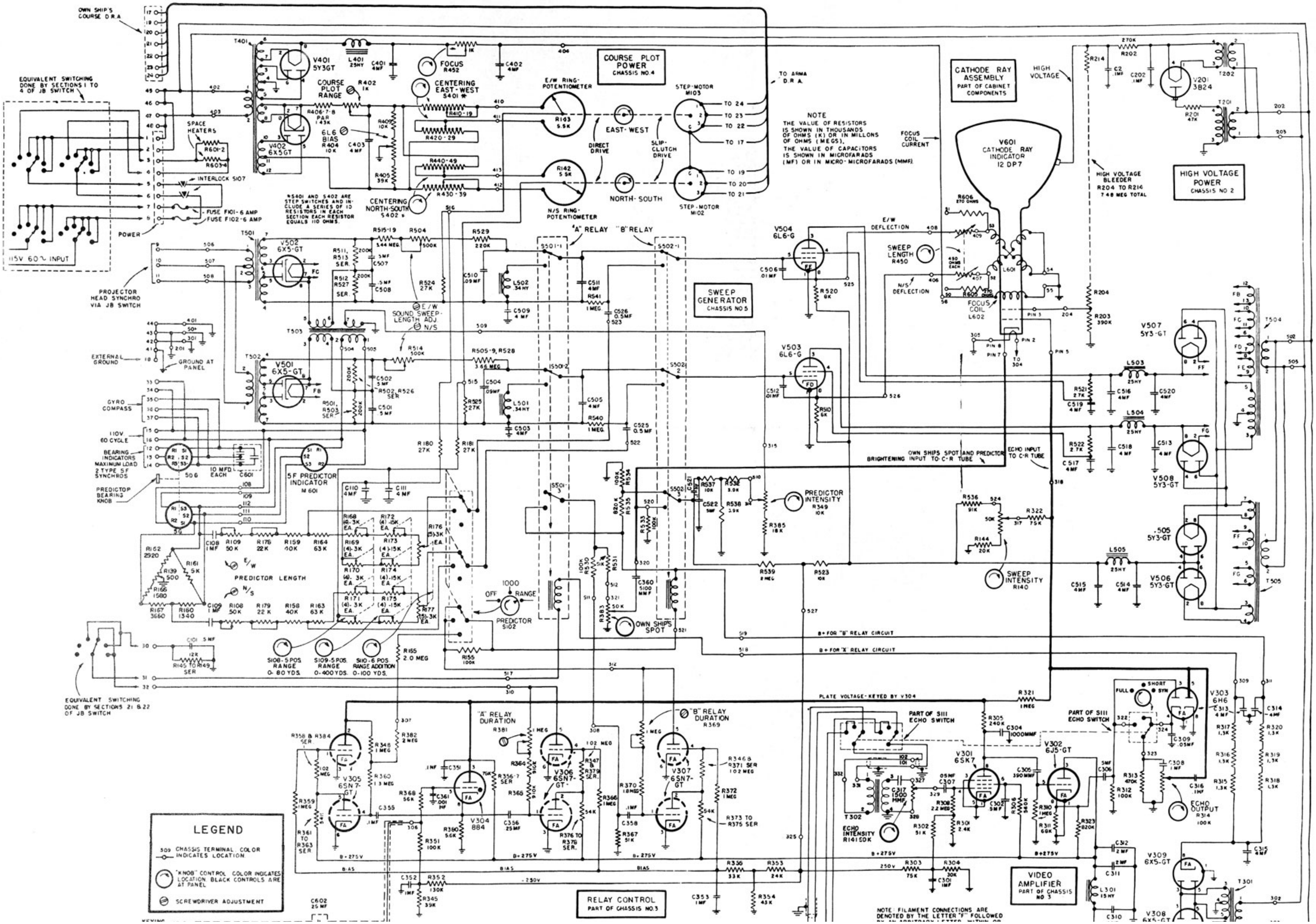
Sonar Home Page

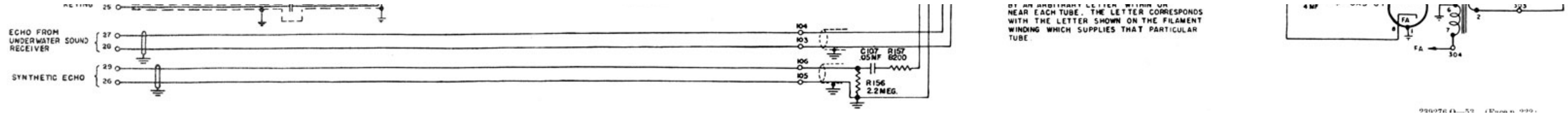


239276 0-53 (Face p. 220)

Figure 11-9 -Circuit schematic of the attack plotter.

[Sonar Home Page]

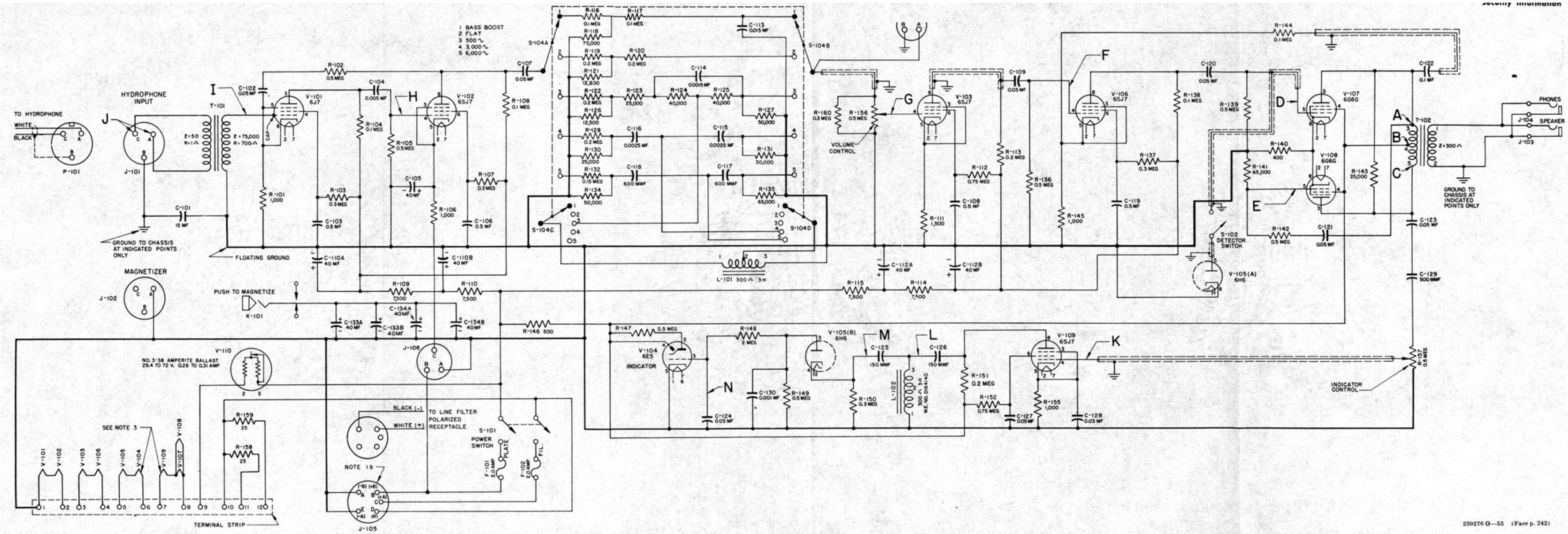




239276 0-53 (Face p. 222)

Figure 13-3 -Circuit of the JP-1 audio amplifier with line filter.

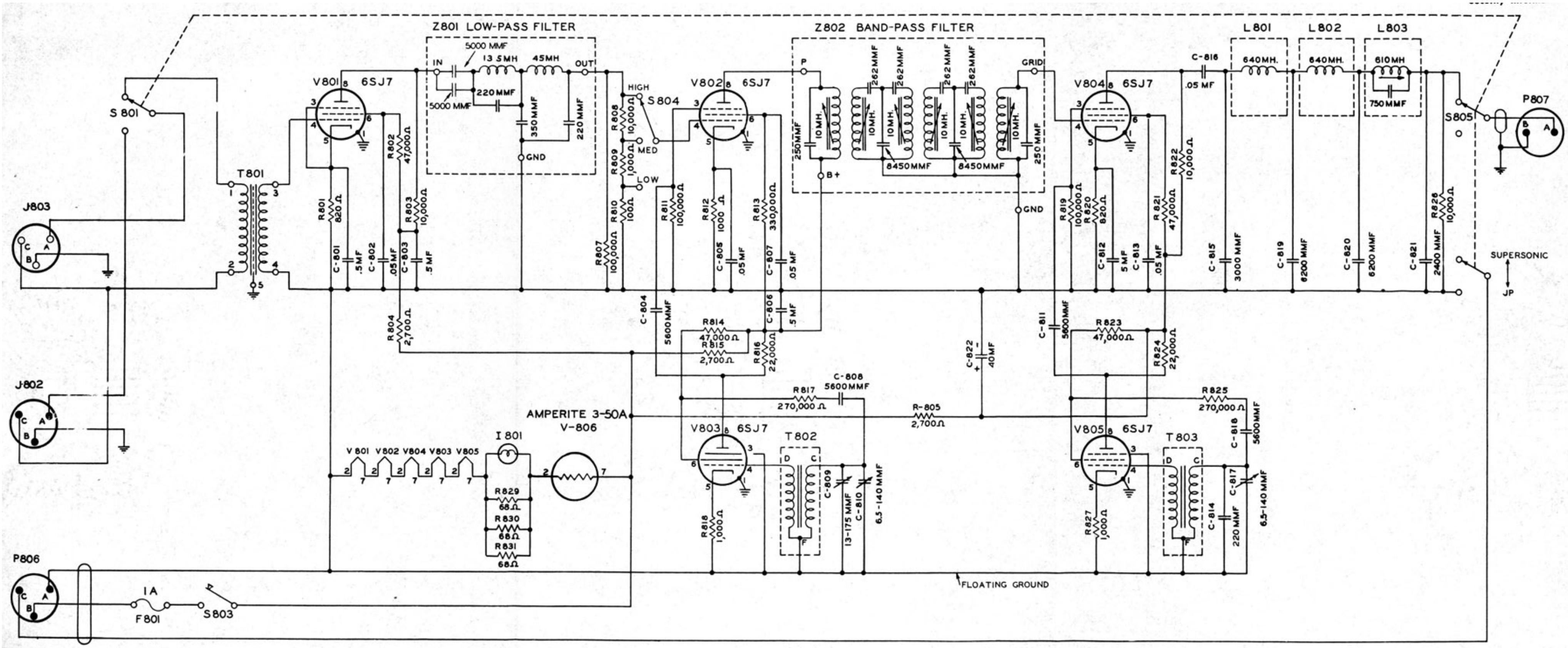
Sonar Home Page



239278 0-53 (Face p. 242)

Figure 13-15 -Schematic diagram of the supersonic-converter unit.

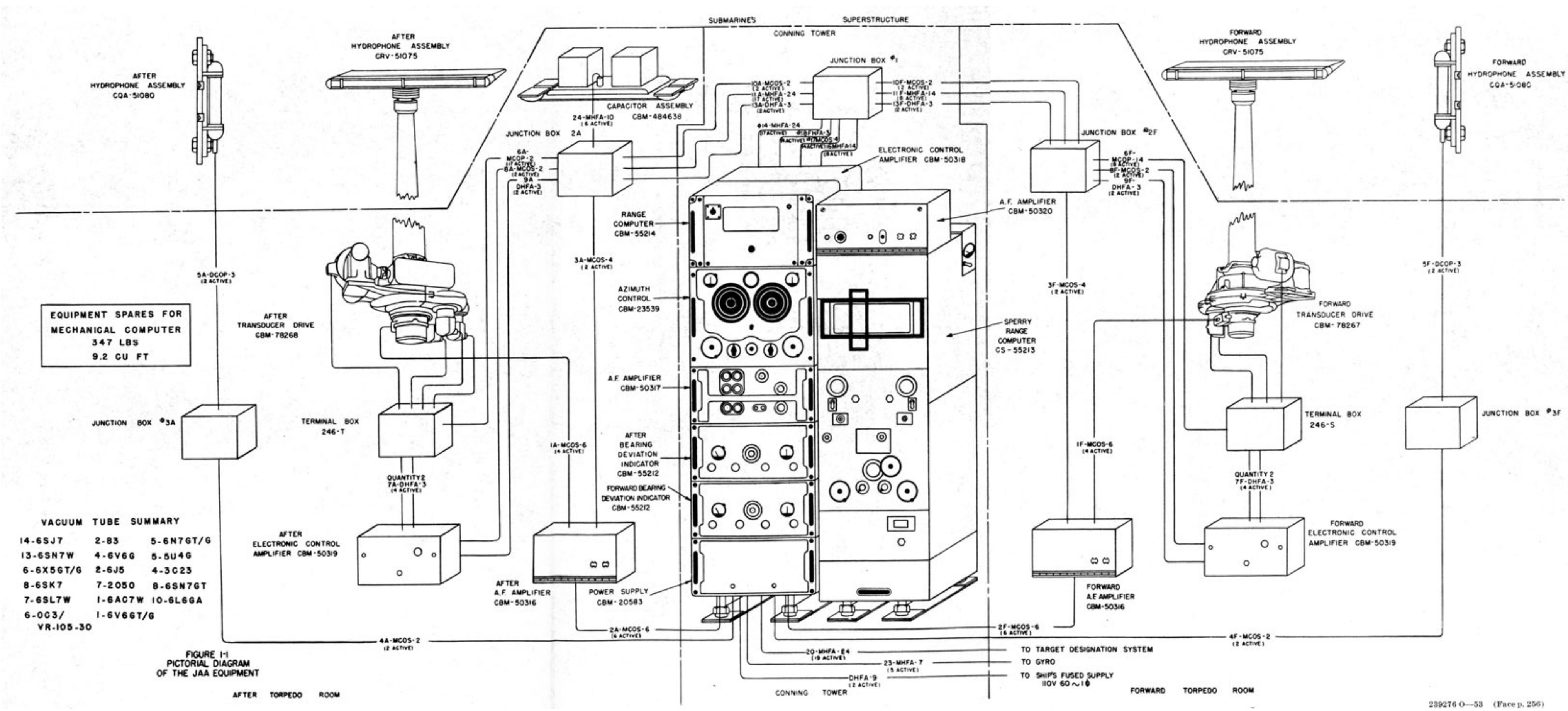
Sonar Home Page



239276 0-53 (Face p. 254)

Figure 13-17 -JAA triangulation-listening-ranging equipment.

[Sonar Home Page]



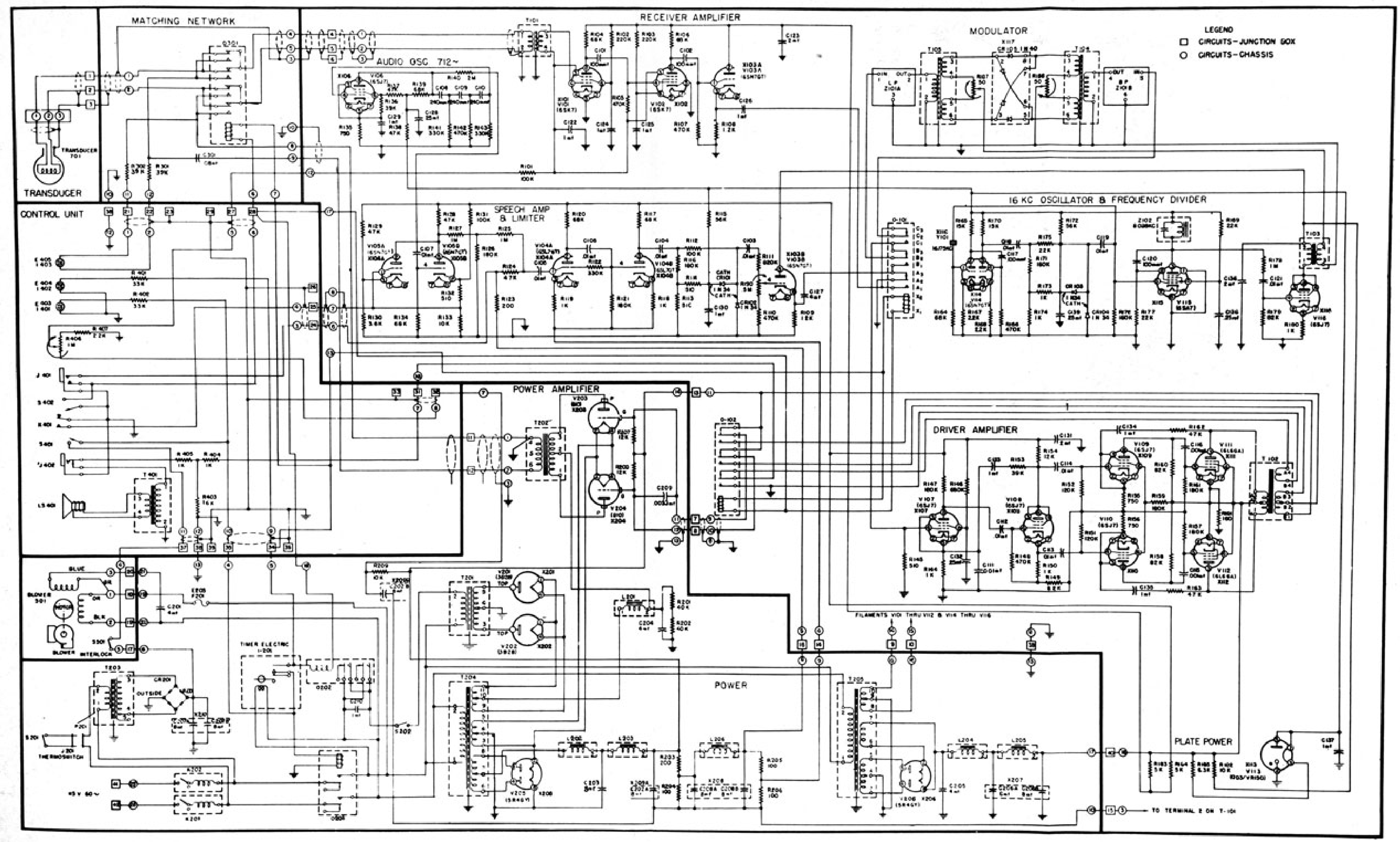
VACUUM TUBE SUMMARY

14-6SJ7	2-83	5-6N7GT/G
13-6SN7W	4-6V6G	5-5U4G
6-6X5GT/G	2-6J5	4-3C23
8-65K7	7-2050	8-65N7GT
7-6SL7W	1-6AC7W I0-6L6GA	
6-003/VR-I05-30	1-6V667/G	

239276 O-53 (Face p. 256)

Figure 13-22. -Schematic diagram of the AN/UQC-1 equipment.

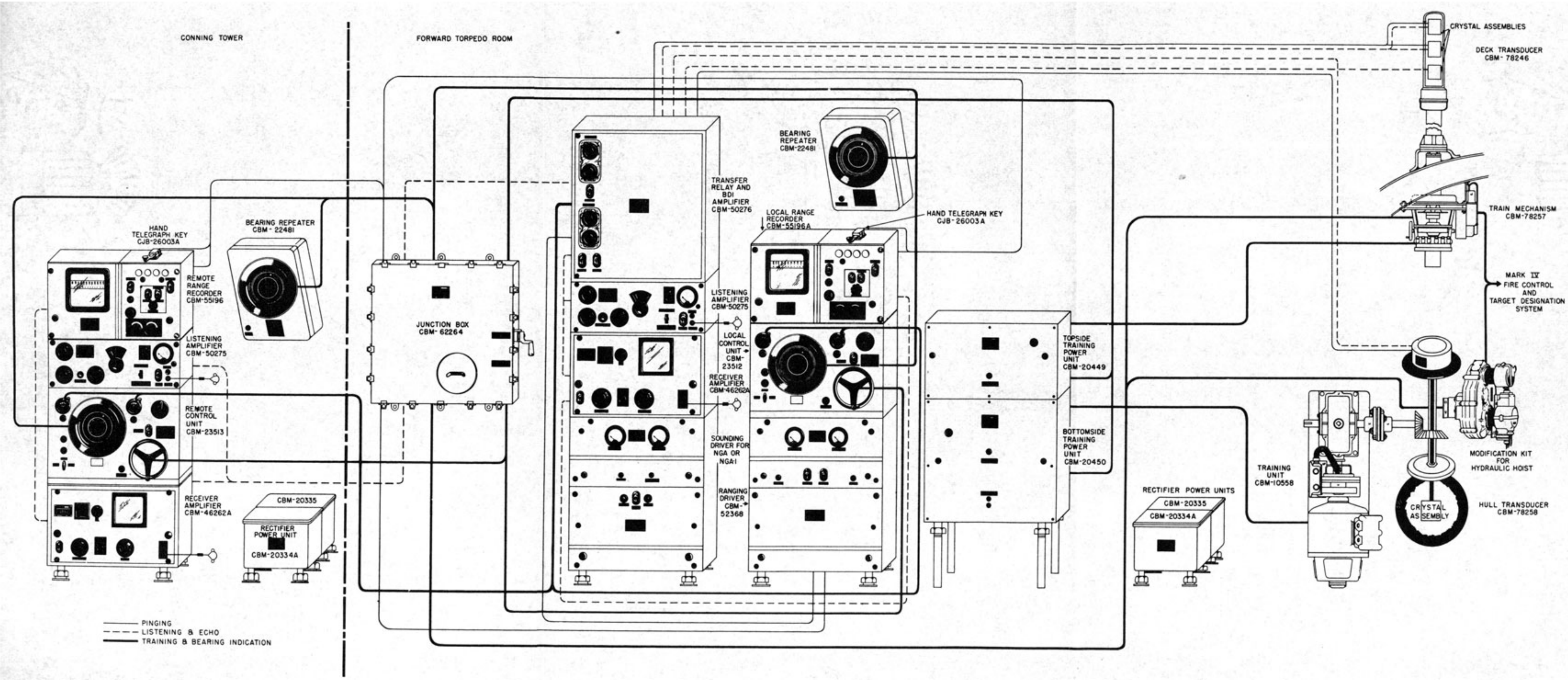
Sonar Home Page



239276 0-53 (Face p. 260)

Figure 14-3. -Over-all pictorial diagram of the WFA-1 equipment.

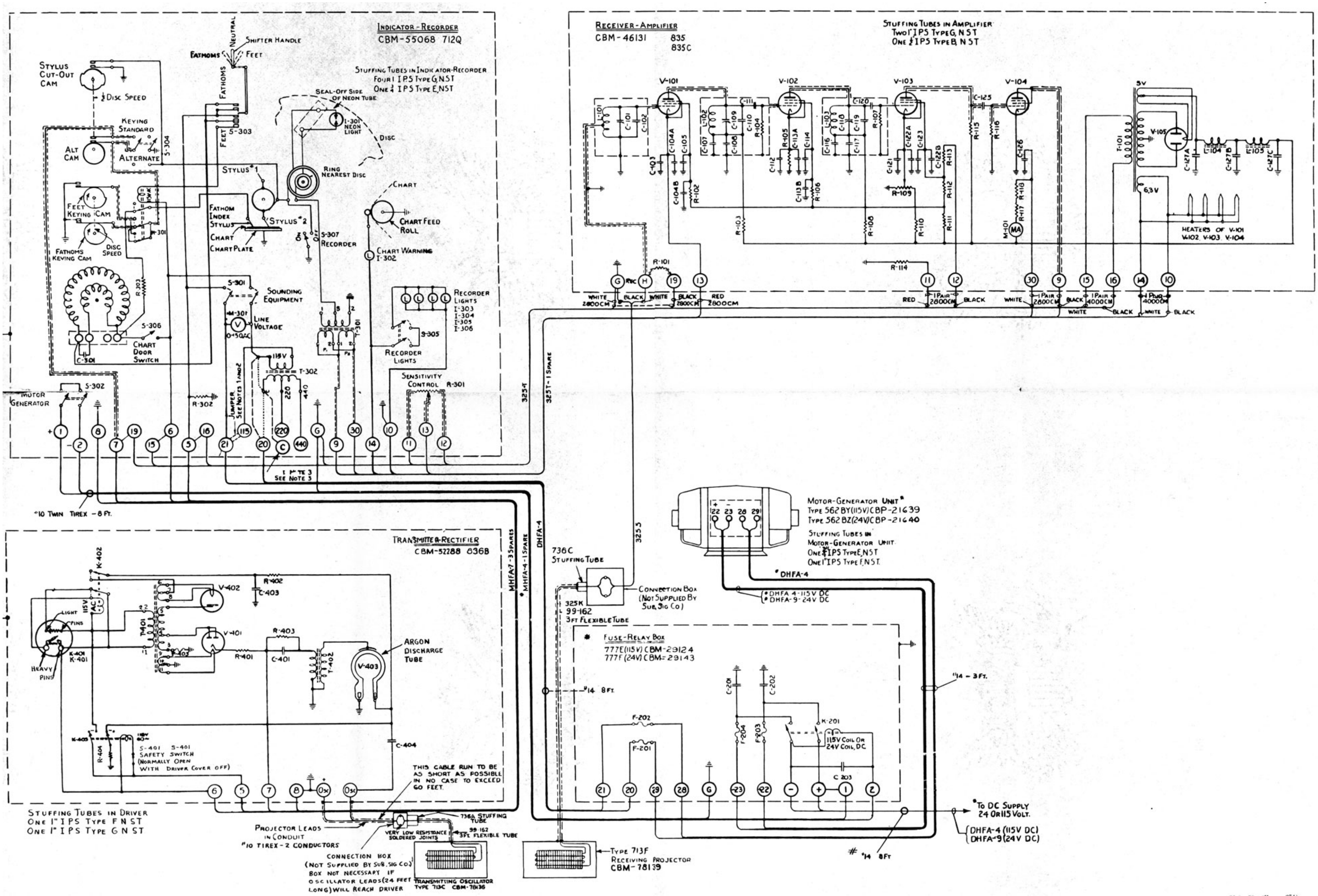
Sonar Home Page



239276 O-53 (Face p. 264)

Figure 15-6. -Schematic wiring diagram of the NJ-9 sounding equipment.

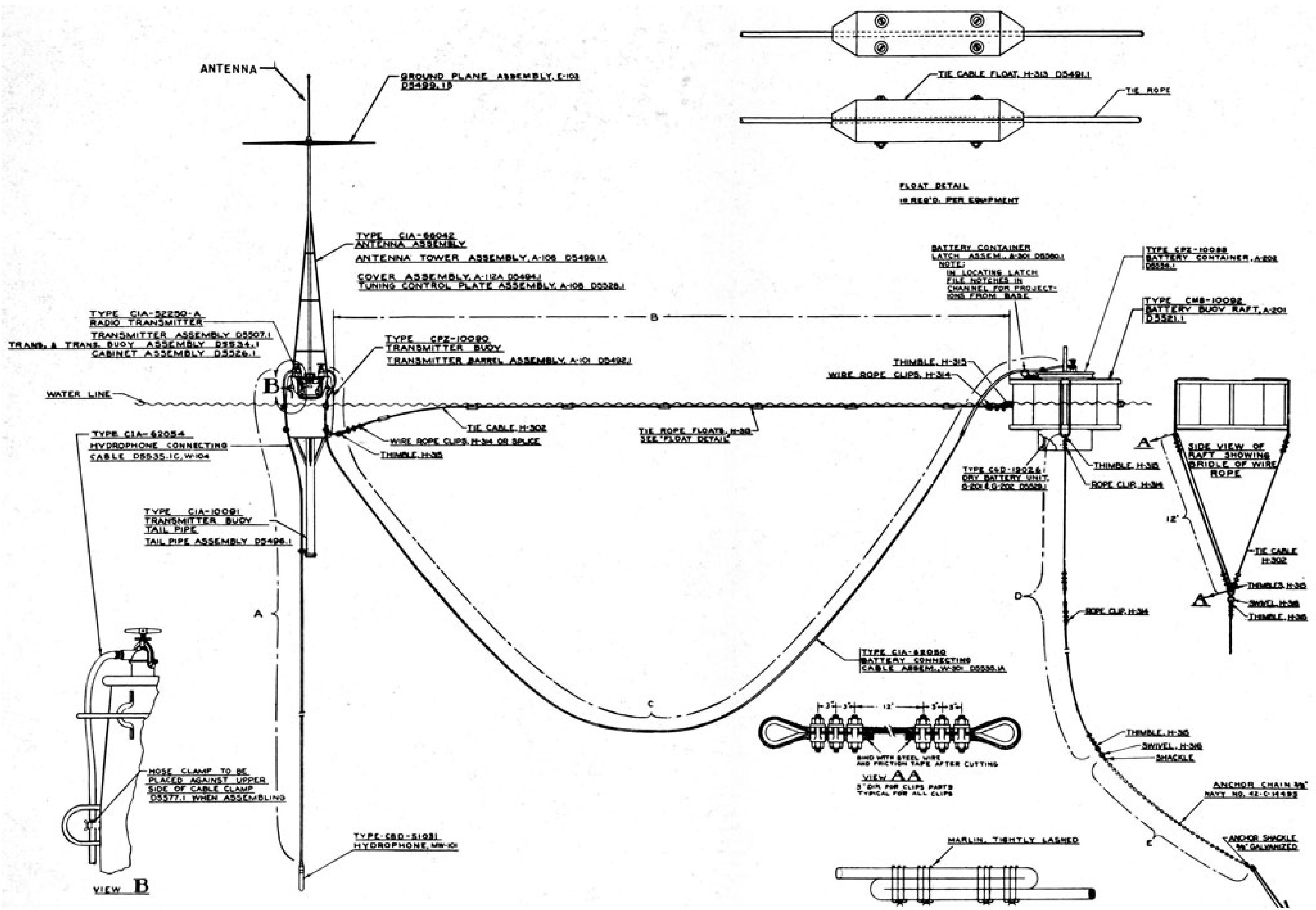
Sonar Home Page



239276 0-53 (Face p. 274)

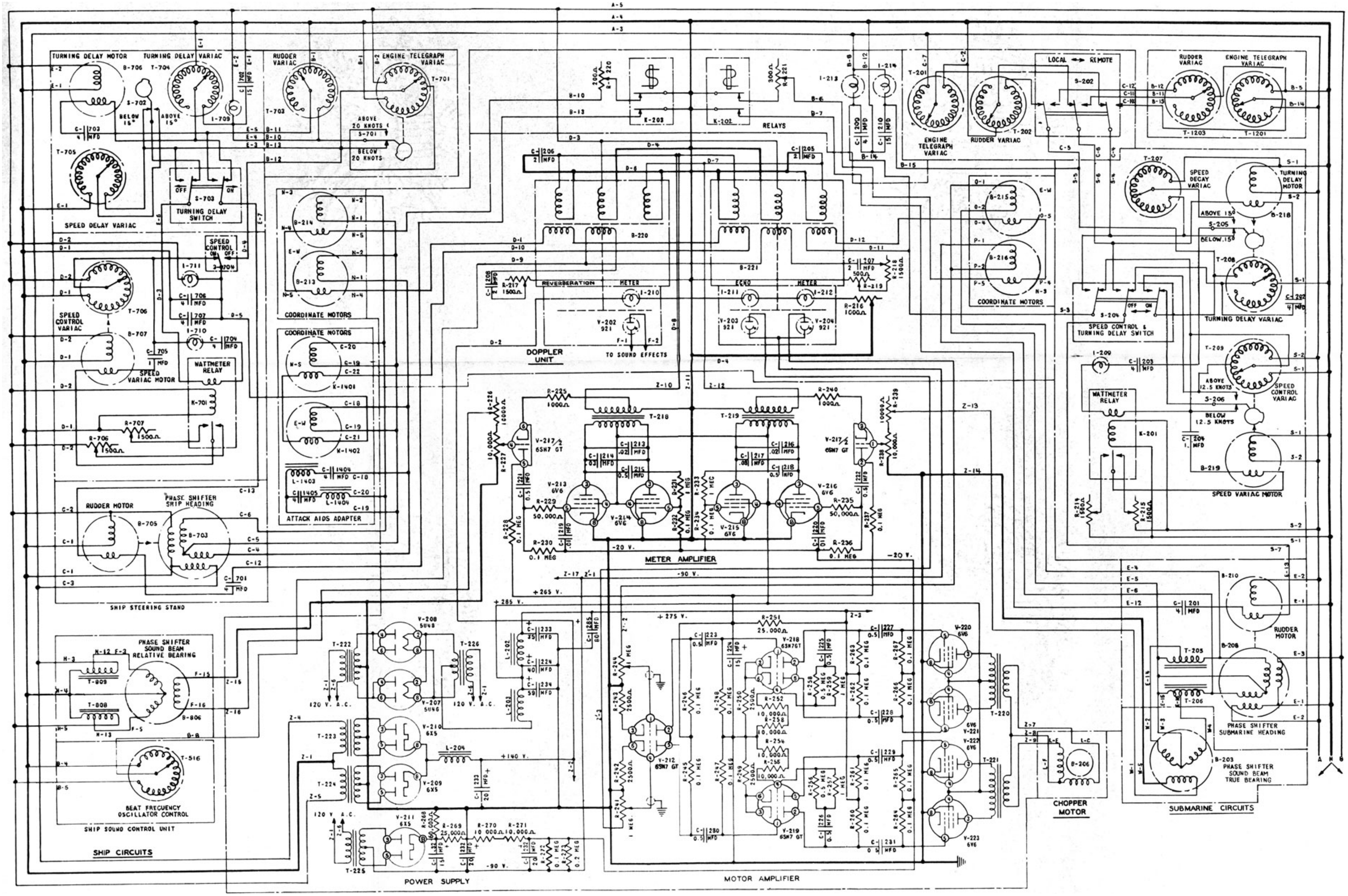
Figure 16-10. -Model JM-4 radio sonobuoy equipment.

Sonar Home Page



239276 0-53 (Face p. 292)

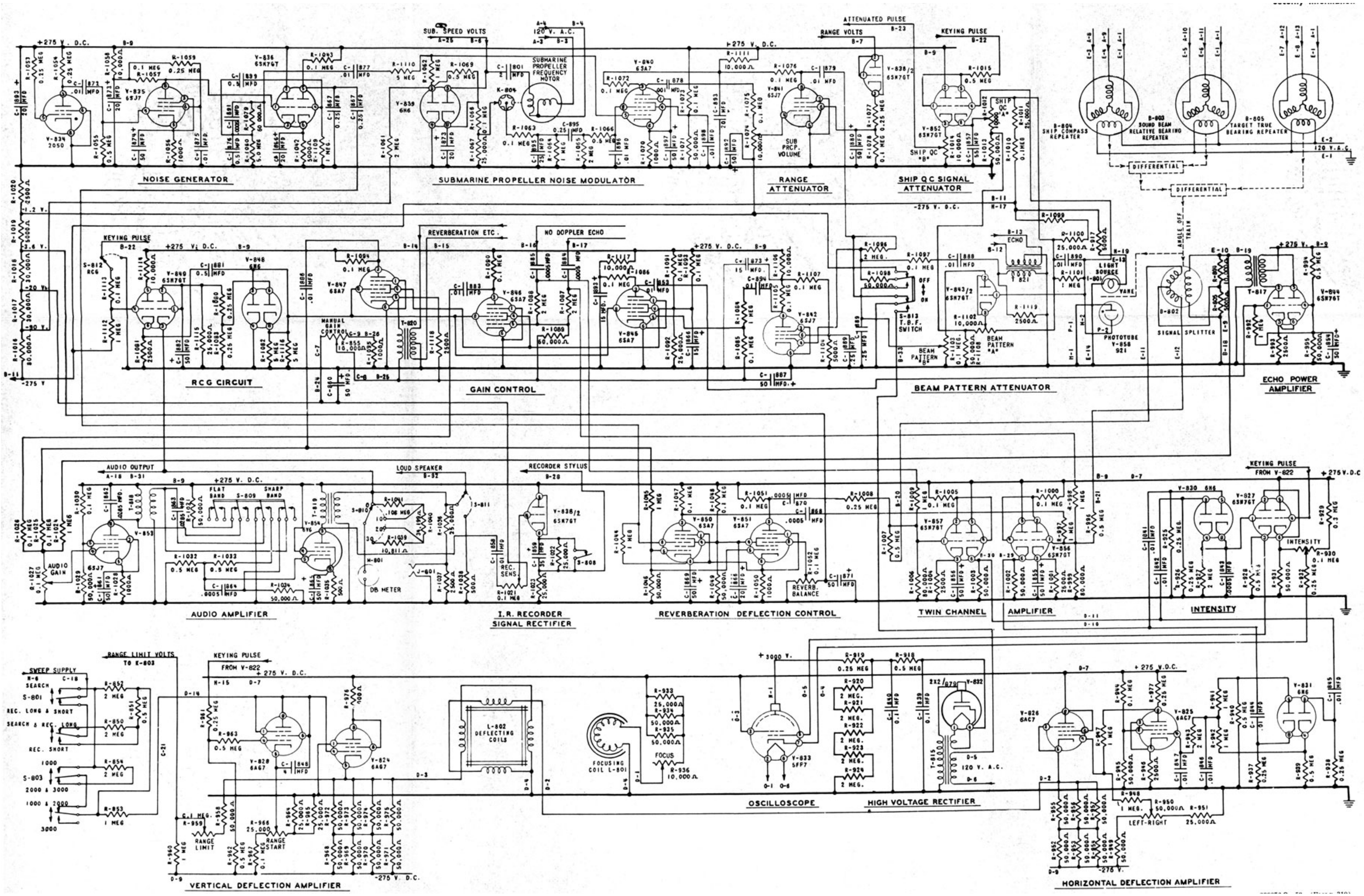
Figure 17-3 -Schematic diagram of speed and direction controls of the attack teacher. Sonar Home Page



239276 0-53 (Face p. 306)

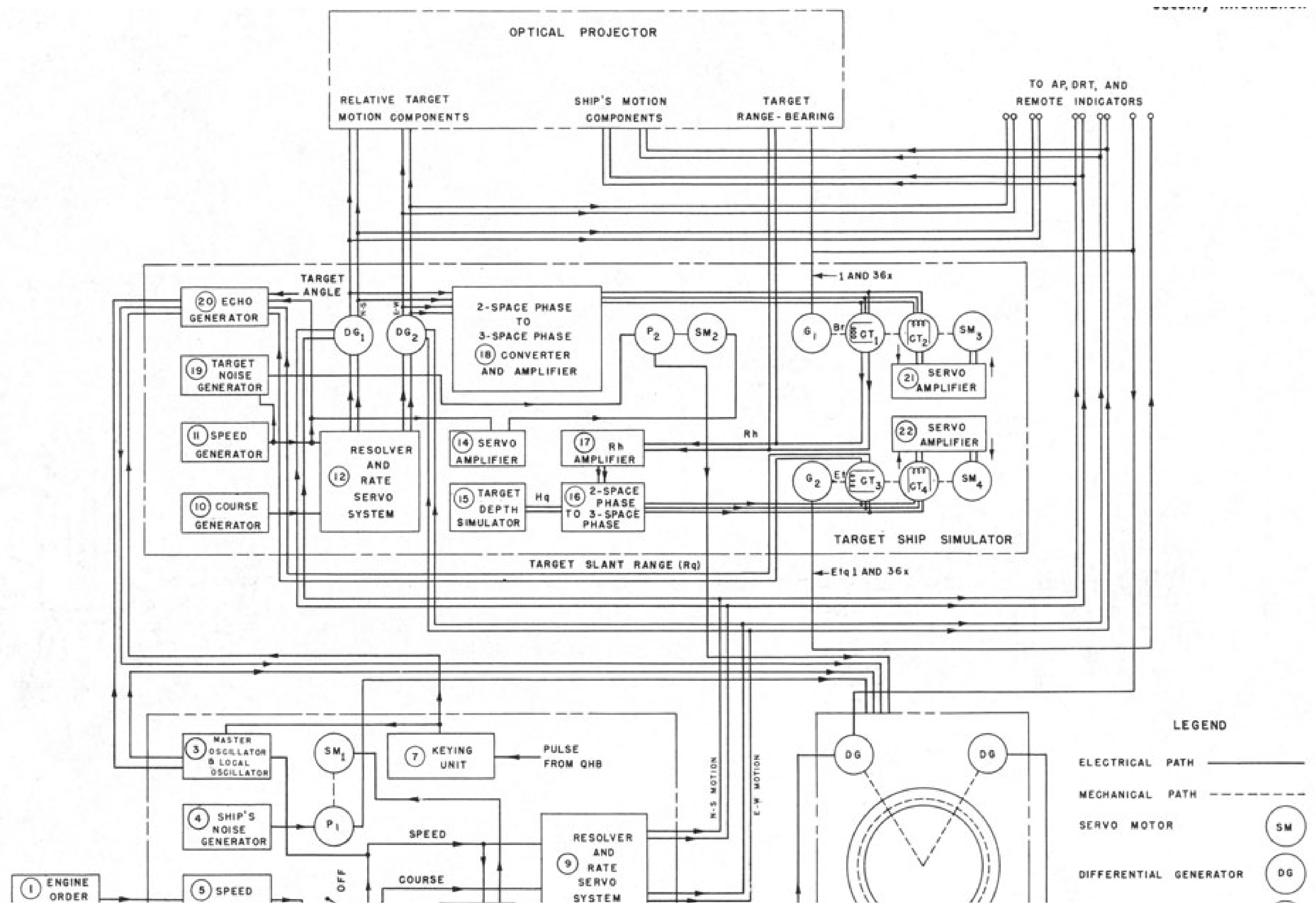
Figure 17-5 -Schematic diagram of the receiver-amplifier and bear-ring-deviation indicator of the QFA-6.

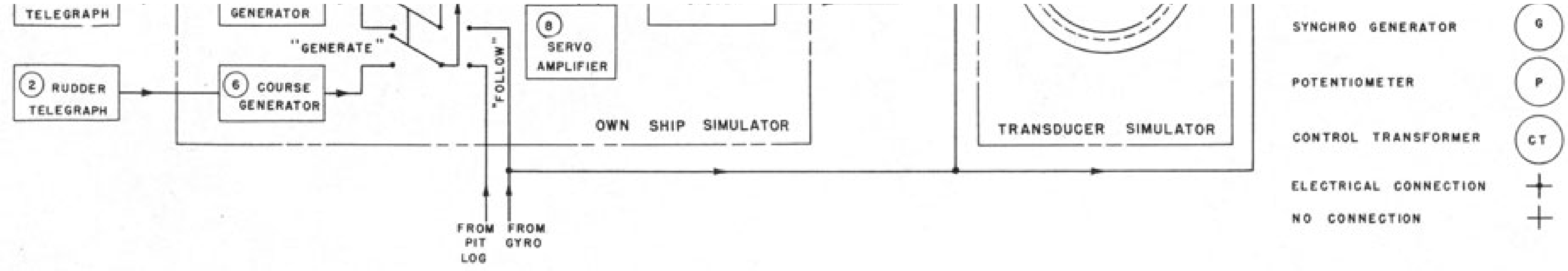
Sonar Home Page



239276 0-53 (Face p. 310)

Figure 17-7 -Simplified functional diagram of AN/UQS-T1. Sonar Home Page





239276 0-53 (Face p. 318)

Historic Naval Ships Association

Historic Naval Ship Visitors' Guide

Historic Naval Ships Listed By Location
Historic Naval Ships Listed By Name
Historic Naval Ships Listed By Type
HNSA Ships with Overnight Programs
HNSA Associate Members

About HNSA
How To Join HNSA
Sources of Information

New-Historical Sound Online

New-WW II Destroyer Operating in 2006

New-Navy Documents and Manuals Online

HNSA Bulletin Board
Standards for Historic Vessel Preservation Projects
Job Postings in the Historic Fleet
Join the next HNSA Conference, 9 Oct 2006 at HMS *Belfast*, London, U. K.

Search hnsa.org

Historic Naval Ships by Location:

AUSTRALIA

HMAS *Ovens*, Fremantle, Western Australia
HMAS *Diamantina*, South Brisbane, Queensland
SS *Forceful*, South Brisbane, Queensland
HMAS *Advance*, Sydney, New South Wales
Commando Boat *Krait*, Sydney, New South Wales
MB172 Epiclass, Sydney, New South Wales
HMAS *Onslow*, Sydney, New South Wales
HMAS *Vampire*, Sydney, New South Wales

HMAS *Whyalla*, Whyalla, South Australia

CANADA

RV *Ben Franklin*, Vancouver, British Columbia

RCMPV *St. Roch*, Vancouver, British Columbia

HMCS *Fraser*, Bridgewater, Nova Scotia

HMCS *Sackville*, Halifax, Nova Scotia

HMCS *Haida*, Hamilton, Ontario

ENGLAND

HMS *Belfast*, London

HMS *Cavalier*, Chatham

FRANCE

FS *Colbert*, Bordeaux

GREECE

B/S *Georgios Averoff*, Athens

ISRAEL

INS *Af Al Pi Chen*, Haifa

INS *Mivtach*, Haifa

NETHERLANDS

HNLMS *Tonijn*, Den Helder

HNLMS *Abraham Crijnssen*, Den Helder

HNLMS *Schorpioen*, Den Helder

Hr. Ms. *Buffel*, Rotterdam

Hr. Ms. *Mercuur*, Scheveningen

NORWAY

M314 *Alta*, Oslo

PERU

BAP *Abtao*, Callao

RUSSIA

Cruiser *Aurora*, St. Petersburg

Soviet *B-413*, Kaliningrad

Icebreaker *Krasin*, St. Petersburg

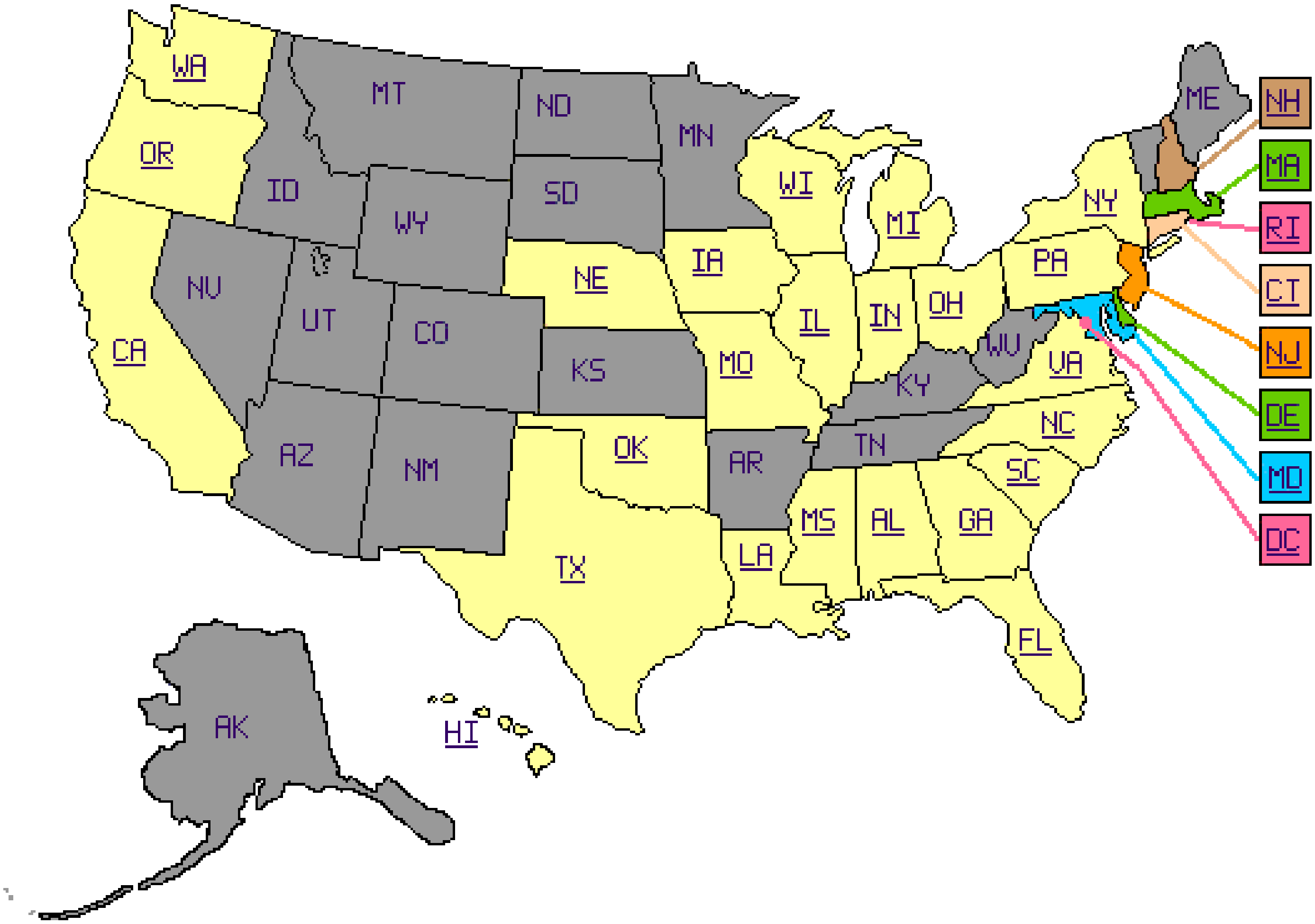
SWEDEN

T121 *Spica*, Stockholm

TURKEY

TCG *Uluçalireis*, Istanbul

UNITED STATES OF AMERICA



[AL] [AR] [CA] [CT] [DC] [DE] [FL] [GA] [HI] [IL] [IN] [IA] [LA] [MD] [MA] [MI] [MS] [MO] [NE] [NH] [NJ] [NY] [NC] [OH] [OK] [OR] [PA] [RI] [SC] [TX] [VA] [WA] [WI]

Alabama

USS *Alabama*, Mobile, Alabama

USS *Drum*, Mobile, Alabama

PBR Mark II, Mobile, Alabama

Arkansas

USS *Razorback*, North Little Rock, Arkansas

California

USS *Hornet*, Alameda, California

USS *Potomac*, Oakland, California
Lightship *Relief*, Oakland, California
SS *Red Oak Victory*, Richmond, California
USCGC Fir, Rio Vista, California
PTF 26, Rio Vista, California
USAT *LT-1967*, San Diego, California
Steam Yacht *Medea*, San Diego, California
USS *Midway*, San Diego, California
SS *Jeremiah O'Brien*, San Francisco, California
USS *Pampanito*, San Francisco, California
SS *Lane Victory*, San Pedro, California
PBR Mark II, Vallejo, California

Connecticut

USCG Boat *Icebucket*, Bridgeport, Connecticut
Japanese *HA-8*, Groton, Connecticut
USS *Nautilus*, Groton, Connecticut
Italian *Siluro a Lenta Corsa*, Groton, Connecticut, USA
USS *X-1*, Groton, Connecticut
Auxilliary Schooner *Brilliant*, Mystic, Connecticut
USCGC *Eagle*, New London, Connecticut

Delaware

Lightship *Overfalls*, Lewes, Delaware

District Of Columbia

USS *Barry*, Washington, District of Columbia
LCVP, Washington, District of Columbia
Motor Whaleboat, Washington, District of Columbia
PCF-1, Washington, District of Columbia
RV *Trieste*, Washington, District of Columbia
Continental Gunboat *Philadelphia*, Washington, District of Columbia

Florida

PTF 3, Deland, Florida, USA
PBR Mark II, Orlando, Florida, USA
SS *American Victory*, Tampa, Florida

Georgia

CSS *Chattahoochee*, Columbus, Georgia
CSS *Jackson*, Columbus, Georgia

Hawaii

- USS *Arizona*, Honolulu, Hawaii
- USS *Bowfin*, Honolulu, Hawaii
- Japanese *Kaiten*, Honolulu, Hawaii
- USS *Missouri*, Honolulu, Hawaii
- USS *Utah*, Honolulu, Hawaii

Illinois

- German *U-505*, Chicago, Illinois

Indiana

- USS *LST-325*, Evansville, Indiana

Iowa

- USACOE Dredge *William M. Black*, Dubuque, Iowa

Louisiana

- USS *Kidd*, Baton Rouge, Louisiana

Maryland

- Lightship *Chesapeake*, Baltimore, Maryland
- USS *Constellation*, Baltimore, Maryland
- SS *John W. Brown*, Baltimore, Maryland
- USCGC *Taney*, Baltimore, Maryland
- USS *Torsk*, Baltimore, Maryland

Massachusetts

- USS *Cassin Young*, Boston, Massachusetts
- USS *Constitution*, Boston, Massachusetts
- Tug *Luna*, Boston, Massachusetts
- *Demolition Boat*, Fall River, Massachusetts
- *Hiddensee*, Fall River, Massachusetts
- USS *Joseph P. Kennedy, Jr*, Fall River, Massachusetts
- *LCM 56*, Fall River, Massachusetts
- USS *Lionfish*, Fall River, Massachusetts
- USS *Massachusetts*, Fall River, Massachusetts
- *PT 617*, Fall River, Massachusetts
- *PT 796*, Fall River, Massachusetts
- USS *Salem*, Quincy, Massachusetts
- German *Seehund*, Quincy, Massachusetts

Michigan

USCGC *Bramble*, Port Huron, Michigan
SS *City Of Milwaukee*, Manistee, Michigan
USCGC *McLane*, Muskegon, Michigan
SS *Milwaukee Clipper*, Muskegon, Michigan
USS *Silversides*, Muskegon, Michigan

Mississippi

USS *Cairo*, Vicksburg, Mississippi

Missouri

USS *Aries*, Brunswick, Missouri

Nebraska

USS *Hazard*, Omaha, Nebraska
USS *Marlin*, Omaha, Nebraska

New Hampshire

USS *Albacore*, Portsmouth, New Hampshire

New Jersey

USS *New Jersey*, Camden, New Jersey
Japanese *Kaiten*, Hackensack, New Jersey
USS *Ling*, Hackensack, New Jersey
PBR Mark II, Hackensack, New Jersey
German *Seehund*, Hackensack, New Jersey
Fenian Ram, Paterson, New Jersey
Holland Boat #1, Paterson, New Jersey
Intelligent Whale, Sea Girt, New Jersey

New York

USS *Slater*, Albany, New York
USS *Croaker*, Buffalo, New York
USS *Little Rock*, Buffalo, New York
PTF 17, Buffalo, New York
USS *The Sullivans*, Buffalo, New York
MV *Commander*, Cornwall-on-Hudson, New York
USS *Growler*, New York, New York
USS *Intrepid*, New York, New York
USAT *LT-5*, Oswego, New York

Admiral's Barge, Romulus, New York

North Carolina

USS *North Carolina*, Wilmington, North Carolina

Ohio

USS *Cod*, Cleveland, Ohio
SS *William G. Mather*, Cleveland, Ohio

Oklahoma

USS *Batfish*, Muskogee, Oklahoma

Oregon

USS *Blueback*, Portland, Oregon
PT-658, Portland, Oregon

Pennsylvania

U.S. Brig *Niagara*, Erie, Pennsylvania
USS *Becuna*, Philadelphia, Pennsylvania
USS *Olympia*, Philadelphia, Pennsylvania
USS *Requin*, Pittsburgh, Pennsylvania

Rhode Island

Soviet *Juliett 484*, Providence, Rhode Island

South Carolina

USCGC *Ingham*, Mount Pleasant, South Carolina
USS *Laffey* Mount Pleasant, South Carolina
USS *Yorktown*, Mount Pleasant, South Carolina
USS *Clamagore*, Mount Pleasant, South Carolina
CSS *H. L. Hunley*, North Charleston, South Carolina

Texas

USS *Lexington*, Corpus Christi, Texas
Admiral's Barge, Fredericksburg, Texas
Japanese *HA-19*, Fredericksburg, Texas
PT 309, Fredericksburg, Texas
USS *Cavalla*, Galveston, Texas
USS *Stewart*, Galveston, Texas
USS *Texas*, LaPorte, Texas
USS *Orleck*, Orange, Texas

Virginia

USS *Monitor*, Newport News, Virginia
USS *Wisconsin*, Norfolk, Virginia
RV *Aluminaut*, Richmond, Virginia

Washington

USS *Turner Joy*, Bremerton, Washington
RV *Deep Quest*, Keyport, Washington
RV *Trieste II*, Keyport, Washington
Tug *Arthur Foss*, Seattle, Washington
Lightship *Swiftsure*, Seattle, Washington
Schooner *Wawona*, Seattle, Washington

Wisconsin

USS *Cobia*, Manitowoc, Wisconsin
USCG Boat *Icelander*, Manitowoc, Wisconsin

Version 2.24, 28 Mar 06

HNSA Web Site Legal Notices

Liability:

The materials on this website are intended to be for informational purposes only.

This Website (excluding linked sites) is controlled by HNSA from its offices within the State of Virginia, United States of America. It can be accessed from all 50 states, as well as from other countries around the world. As each of these jurisdictions has laws and regulations that may differ from those of the State of Virginia, by accessing this Website both you and HNSA agree that the statutes and laws of the State of Virginia, without regard to conflicts of law principles thereof, will apply to all matters relating to use of this website. In the case of a dispute, you and HNSA agree to submit to the exclusive personal jurisdiction and venue of the Superior Court of the County of Isle of Wight, Virginia and the United States District Court for Virginia with respect to such dispute.

HNSA assumes no liability for the use or interpretation of information contained herein. This publication is provided "AS IS" WITHOUT WARRANTY OF ANY KIND, EITHER EXPRESSED OR IMPLIED, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY, FITNESS FOR A PARTICULAR PURPOSE, OR NON-INFRINGEMENT OF THIRD PARTY RIGHTS.

This website contains links to other resources on the Internet including our members, affiliates and others. These links are provided solely as aids to assist you in locating other Internet resources that may be of interest. They are not intended to state or imply that HSNA sponsors, endorses, is affiliated or associated with such linked sites. We do not control and cannot guaranty the relevance, timeliness, legality, cost, accuracy or any other properties of sites linked from our site.

Privacy:

Our computer system uses software programs to create summary statistics that are used for determining the volume of visitors, sources of referrals, determination of other system performance areas, errors in access, and assessing what information is of most or least interest. HNSA collects and stores the following information, The name of the domain from which you access the Internet; The date and time you access our web site; The pages you peruse and files you access; The Internet address of the web site from which you linked directly to our site.

If you choose to provide us with personal information, as in an email message or a web services request form, HSNA will use this information to respond to your request. There are times when your email may be forwarded to others both in and out of HNSA to better assist you. Except for the exceptions listed above and for authorized law enforcement activities, HNSA does not share email or other personal information with outside individuals or organizations without obtaining your permission.

Questions, comments and other communications that we receive may become part of the permanent record of HNSA.

You may send us e-mail. However you should not send us confidential or sensitive information via e-mail because the security of Internet-based e-mail is uncertain. By sending unencrypted e-mail messages containing sensitive or confidential information, you accept the risks of such uncertainty and possible lack of confidentiality over the Internet.

Copyright:

This web site is protected by copyright law and international treaties. Unauthorized reproduction or distribution of the content of this web site, or any portion thereof, may result in severe criminal or civil penalties and will be prosecuted to the maximum extent possible under the law.

Private Use: Contents of this web site may be used for personal and/or educational purposes without restriction

Commercial Use: Contents of this web site, or any portion thereof, including but not limited to non-public domain images, may not be used for any commercial purpose without specific authorization from HNSA.

The HNSA logo and HNSA crest are trademarks of HNSA.

Photos, drawings, and text in the ship pages may also be the intellectual property of others and is subject to their restrictions.

Disputes:

Should you have questions or disputes about these policies, please contact us with a detailed description of your question or dispute at contact below.

Contact Information:

Director
Historic Naval Ships Association
Post Office Box 401
Smithfield, VA 23431-0401
Tel: (757) 356-9422
Fax: (757) 356-9433
Email: info@hnsa.org

Return to the HNSA Guide.

Version 1.06, 10 Jan 06